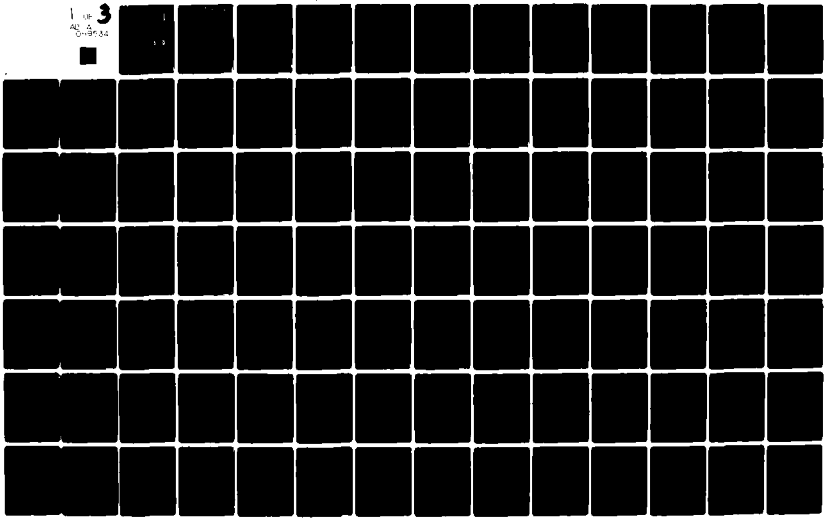
AD-A089 534 ARMY COMMAND AND GENERAL STAFF COLL FORT LEAVENWORTH KS F/G 17/2
CONVERSION OF THE DEFENSE COMMUNICATIONS SYSTEM FROM ANALOG TO --ETC(U)
JUN 74 R A STANLEY
UNCLASSIFIED SBIE-AD-E750 001 NL

1 OF 3
AD A
089534



PHOTOGRAPH THIS SHEET

AD-E250 001

AD A089534

DTIC ACCESSION NUMBER

III

LEVEL

INVENTORY

Army Command & General Staff College, Ft. Leavenworth, KS

Conversion of the Defense Communications System from Analog to Digital Form

DOCUMENT IDENTIFICATION

Stanley, Richard A. June 1974

DISTRIBUTION STATEMENT A

Approved for public release;
Distribution Unlimited

DISTRIBUTION STATEMENT

ACCESSION FOR		
NTIS	GRA&I	☒
DTIC	TAB	☐
UNANNOUNCED		☐
JUSTIFICATION		
BY Per Ltr. on file		
DISTRIBUTION /		
AVAILABILITY CODES		
DIST	AVAIL AND/OR SPECIAL	
A		

DISTRIBUTION STAMP

DTIC
ELECTE
SEP 24 1980
S D
D

DATE ACCESSIONED

80 9 10 064

DATE RECEIVED IN DTIC

PHOTOGRAPH THIS SHEET AND RETURN TO DTIC-DDA-2

DTIC FORM OCT 79 70A

DOCUMENT PROCESSING SHEET

N-19052.174-2

UNCLASSIFIED

AD A089534

UNCLASSIFIED

N-19052.174-2

AD-E750001

6

CONVERSION OF THE DEFENSE COMMUNICATIONS SYSTEM

FROM ANALOG TO DIGITAL FORM.

A thesis presented to the Faculty of the U.S. Army Command and General Staff College in partial fulfillment of the requirements of the degree

MASTER OF MILITARY ART AND SCIENCE

by

10 RICHARD A. STANLEY, CPT, USA
BSEE, Lehigh University, 1963
MS, Lehigh University, 1964

Fort Leavenworth, Kansas
1974

THESIS APPROVAL PAGE

Name of Candidate Richard A. Stanley, CPT, USA

Title of Thesis Conversion of the Defense Communications System from Analog to Digital Form

Approved by:

F. Putnam, LTC, SC, Research and Thesis Adviser

Elmer A. Brown, LTC, SC, Member, Graduate Research Faculty

John Steinmeier, LTC, SC, Member, Graduate Research Faculty

J. G. Rosenbarker, LTC, SC, Member, Consulting Faculty

Ronald F. Heiden, LTC, IN, Member, Consulting Faculty

Date: 4 June 1974

The opinions and conclusions expressed herein are those of the individual student author and do not necessarily represent the views of either the U.S. Army Command and General Staff College or any other governmental agency. *(References to this study should include the foregoing statement.)*

SECURITY STATEMENT

Although certain sources cited in the bibliography of this paper bear protective security markings, nothing of a classified nature has been extracted from those documents either directly or by inference. This paper contains no classified material and needs neither protective markings nor handling.

ABSTRACT

CONVERSION OF THE DEFENSE COMMUNICATIONS SYSTEM
FROM ANALOG TO DIGITAL FORM

CPT Richard A. Stanley, June 1974, Pp. ix+190

Given that the Defense Communications System (DCS) plans to convert from analog to digital transmission, the paper examines that decision in an attempt to determine major problem areas and topics worthy of further study. The DCS history, organization, and structure are examined, and modulation theory and techniques are reviewed. Current development in advanced communications technology is examined, and the implications of these developments on the DCS conversion are examined and evaluated. Anticipated major problem areas are deduced, and areas that require further investigation are enumerated.

ACKNOWLEDGMENTS

The author wishes to thank the Thesis Committee for its assistance, encouragement, and constructive criticism. Especial thanks are extended to LTC's Putnam, Rosenbarker, and Heiden for the many hours they have spent assisting the writer during the preparation of this paper.

The assistance of CPT Kevin B. Deasy, USA, in obtaining data and documents pertinent to the Defense Communications System and of Mrs. Evelyn F. Randolph in typing the final manuscript are gratefully acknowledged.

Finally and most significantly, the author thanks his wife for her encouragement, patience, and assistance throughout the year, without which this paper could not have been written.

R. A. S.

TABLE OF CONTENTS

LIST OF FIGURES

CHAPTER I

INTRODUCTION

BACKGROUND

The Defense Communications Agency (DCA) has announced its intent to convert the Defense Communications System (DCS) from its present analog form to an all-digital system.[1] The reasons for this decision are threefold:[2]

1. Increased requirement for data traffic capacity in the DCS.

2. The requirement for expanded secure voice service.

3. The requirement for improved signal quality over longer built-up connections than previously used.

The magnitude becomes more apparent when some statistics are considered. The rate of increase of data traffic in the DCS in the past decade has been in the vicinity of 10 per cent per annum.[3] A recent study prepared for the United States Air Force (USAF) shows that greater than 75 per cent of current USAF communications programs require data transmission over long distances in various forms.[4] Congressional investigation of the DCS concluded that the

lack of adequate secure voice capability is the system's most serious fault.[5] DCA has redefined its reference circuit, the standard against which performance is measured, as being 12,000 miles, whereas previously it had been half that.[6]

These phenomena have affected civil networks as well as the DCS. Data traffic growth has been literally explosive in virtually every civilized country of the world.[7] In the United States, the Bell System has used digitized-voice carrier equipment in its exchange plant for more than 10 years.[8] Because of the excellent results its use is growing at a rate of more than 16,000 channel-kilometres per day, while use of analog multiplex systems declines.[9]

NEED FOR AND PURPOSE OF THIS STUDY

In spite of the above, the conversion of an entire communications system of the magnitude of the DCS from analog to digital form in its entirety is without precedent. It is a task of enormous complexity and requires the coordinated application of technical, economic, and social skills, many of which have never been previously tested in practice.

Many studies have been performed on this conversion. Owing to the sheer magnitude of the problem, none is comprehensive and this paper will be no exception. However,

the primary emphasis of previous studies has been either technical or economic, orientated primarily toward the actual conversion of the system, with little attention to the implications of piecemeal system conversion. Similarly, these studies have been directed at experts in the field by experts in the field and have been of little use to the manager who must implement the system conversion.

Therefore, the purposes of this study are as follows:

1. To review the theory of modulation, digital and analog, and examine the comparative effect of transmission impairments upon the signal.

2. To present the major areas of research in the field of data communications, to include the present status and findings of that research.

3. To evaluate the DCS by means of a model, using the knowledge described above and the functional limitations of the system in the next decade, in an attempt to determine major problems that may be expected to arise during conversion of the DCS to digital form.

4. To determine, if possible, whether there are other directions in research that might be beneficial to this conversion project.

LIMITATIONS

This subject is of such magnitude that no single study can hope to cover it in its entirety. In addition to the limitations implicit in the statement of purposes, the study is further delimited as follows:

1. Only the DCS backbone is considered, and that for the period 1975-1985.

2. The paper is system-oriented and primarily qualitative.

3. In the interest of widest dissemination of the paper, potentially classified topics, such as survivability and electronic countermeasures vulnerability, are not discussed.

4. The reader is assumed conversant with basic communications theory and economic analysis.

ORGANIZATION

To accomplish the stated purposes, Chapter II reviews the organization and functions of the DCS, comments on the shortcomings of the system, and concludes with the presentation of a topographical model of the DCS which is used to evaluate the impact on the system of the digital conversion. Although simplistic, the model considers the time- and frequency-dependence of the channel transfer

functions and allows for treatment of alternate routing, tandem connections, and other standard configurations of the DCS.

Modulation theory and the relative effect of transmission impairments on signals are covered in Chapter III. Chapter IV reviews advanced communications technology and its application to practical systems.

Chapter V analyzes the effects of digitization on the DCS and suggests approaches to this problem. A short summary and recommendations for future research are presented in Chapter VI.

CHAPTER II

DEFENSE COMMUNICATIONS SYSTEM

BACKGROUND

The Defense Communications System (DCS) is the worldwide Department of Defense (DOD) telecommunications network and is managed under the single-manager concept by the Defense Communications Agency (DCA), an arrangement that has existed since 1960. The DCA is responsible for the operational direction of the DCS but does not operate it. This task is performed by the three armed services as operating and maintenance agencies. The services are responsible for the funding, installation, and manning of those DCS facilities they operate for the DCA, an arrangement that is the cause of considerable friction between the two levels of command and one that also demands active participation and cooperation between these parties.[1]

DCS ORGANIZATION

The DCS is primarily a common-user network, built largely from the separate service networks which preceded it and which still provide its backbone. Several dedicated

systems transit the DCS (e.g., Worldwide Military Command and Control System), but it remains basically orientated toward the provision of common-user service through one or more of its three major subsystems: AUTODIN, AUTOVON, and AUTOSEVOCOM.

The Automatic Digital Network (AUTODIN) handles 95 per cent of DOD record traffic at speeds ranging from 50 to 4800 baud. It utilizes 19 store-and-forward automatic digital message switches to provide rapid routing of messages worldwide, by precedence, and without manual relay or intervention. The present message volume in AUTODIN is in excess of one million messages daily, and it provides total encryption of the message (to include the address and header) and traffic-flow security.[2]

The Automatic Voice Network (AUTOVON) is a non-secure worldwide telephone network which can be thought of as DOD's direct distance dialing system. By means of 84 switching centers worldwide, it provides direct interconnection of subscribers by circuit switching. AUTOVON incorporates a precedence system to permit the preemption of a low precedence call by one of higher precedence, and the selection of precedences is under user control. A shortage of funds has made AUTOVON access-line limited.[3] Coupled with the ability of the user to select a call precedence

which will preempt all calls of a lower precedence, this limitation has resulted in a call completion rate of only 45 per cent on a global basis,[4] as compared to approximately 75 per cent in commercial networks.[5] This shortcoming results primarily from the fact that AUTOVON, although designed as a command and control system, has become widely used as an administrative network serving a large user population.[6]

The third major subsystem of the DCS is the Automatic Secure Voice System (AUTOSEVOCOM), which provides secure voice service to selected subscribers worldwide.[7] This is a system in name only. It actually consists of several separately conceived and implemented secure voice systems which are connected together through appropriate interfaces to provide a network. This results in a secure voice system that is less than optimum on a system basis, although there are subsystems that work quite satisfactorily. Owing to security limitations, discussion of secure voice networks is quite limited. It is sufficient to realize that such a system exists, that it is--at the moment, at least--rather a miscellany of black boxes,[8] and that it services only *selected* subscribers.

THE DCS TODAY

Today's DCS is primarily a voice-orientated, analog transmission, circuit-switched network. In 1970 the DCS devoted approximately 84 per cent of its circuity to voice traffic, with the remaining 16 per cent devoted to data transmission of various sorts, including AUTOSEVOCOM.[9] These percentages are constantly changing, of course, as the various means of communication grow. The present growth rate for voice communications is in the range of 2 to 8 per cent per annum.[10] Unhappily, the growth rate of data traffic is subject to much more disagreement, with estimates ranging from a continued growth of 10 to 11 per cent per annum[11] to estimates of 25 to 30 per cent.[12] Such a range of possibilities obviously offers the system designer many problems in attempting to design improvements and traffic-handling capabilities for the system.

The primary means of transmission over the DCS backbone are cable, line-of-sight (LOS) microwave, tropospheric scatter radio, satellite, and HF radio, each medium utilized to a different extent in varying sections of the DCS.[13] Thirty-eight per cent of these transmission paths are government-owned and operated. The remaining 62 per cent are leased from domestic and foreign commercial carriers.[14]

The DCS is not completely analogous to civilian networks such as the Bell System for a number of reasons. One of these is the traffic mix in DCS, which is more data-heavy than that of its civilian counterparts. Another variance is the typical channel. Comprising the DCS are 60,000 circuits that traverse more than 45 million channel miles.[15] These data reveal a mean circuit length of 750+ miles, considerably longer than the mean of 10 to 20 miles found in typical commercial networks.[16]

Switching in the DCS falls into two categories: circuit switching, used everywhere except in AUTODIN (it is available in the Continental United States AUTODIN to interconnect high-volume subscribers, but is regarded as a secondary means), and message switching, used in AUTODIN. There are 84 AUTOVON, 135 AUTOSEVOCOM, and 19 AUTODIN switches in the DCS.[17] All are computer-controlled, 4-wire machines, with the circuit switches using space-diversity switching. Regardless of network, all present an analog signal interface to the DCS.

THE DCS, 1980-1985

The future development of the DCS is not an area in which wide consensus exists. The future scope and direction of military communications is very much affected by

political events, the impact of which can be only subjectively evaluated, with questionable precision. Furthermore, the prediction of future trends in any area rests in large part on the projection of past experience. Such projections implicitly assume gradual and predictable change, an assumption that is not necessarily valid in a directed network.

An overview of the DCS of the next decade is developed within the next several pages. The projections are based upon assumptions that appear reasonable and feasible as this is written and should be examined in that context as actual events modify the basis upon which the assumptions were made.

TRANSMISSION MEDIA

It is not anticipated that there will be any quantum technological advances in the area of signal transmission in the next decade. Rather, advances are expected to result from refinement and evolution of technology already known.[18]

The DCA has announced its intention to place increased emphasis upon intercontinental cable within the next decade, with increased use of satellite trunking for flexibility in rerouting capability and circuit engineering. Although the use of satellite circuits will increase markedly, it will still remain a relatively small percentage of

the total DCS channels.[19]

HF radio, already in only limited use in the DCS backbone, will virtually disappear by 1985 as circuits continue to be transferred to other media. This trend will probably not extend into the tactical area, which may create some interfacing problems for the DCS, but these should be minor.

The primary new means of transmission in the 1980's is expected to be millimetre-wave transmission, both in wave-guide and through the atmosphere. The use of millimetre-waves will initially occur in high-density trunking areas and over relatively short distances. However, the extremely large bandwidths obtainable with this means will enable it to achieve economies by displacement of other systems on a greater than one-for-one basis.[20]

Optical transmission, although approaching practicality sooner than originally predicted, still faces many technical problems before it becomes practical for large-scale military use. It may be used experimentally in the DCS by 1985, but it is not a likely contender for large use much prior to 1990.[21]

In 1985, as today, the backbone of the DCS will continue to be line-of-sight microwave radio and tropospheric scatter radio, with rather less of the latter than

is in use today, owing to redeployment of forces from areas served by tropo systems and because of the technical characteristics of tropo. The reasons for this are twofold: the large capital investment in this equipment at present, coupled with its long economic lifetime; and its suitability to all sorts of signal transmission with high accuracy and reliability.

DCA OPERATIONS

The progenitor of DCA and the DCS was economics. Both were created as much to eliminate duplication of facilities and operations as to provide a single line of command for military communications. The situation in 1960 was brought to a head by Congressional interest in the cost of military communications. Much the same situation obtains today.

A Congressional investigation of the Department of Defense communications was recently completed. The study was prompted by a series of communications mishaps (USS *Pueblo*, USS *Liberty*, EC-121 Incident), but its scope extended well beyond the circumstances of those incidents. Among the more important findings and conclusions of the investigation were:[22]

1. DOD communications are poorly managed, as

evidenced by an inability to properly account for operating costs, division of management functions within the DCS, and excessive message handling times.

2. The unresponsiveness of the DCS was a significant factor in the capture of the USS *Pueblo*, the attack on the USS *Liberty*, and the shooting down of an EC-121 over Korea.

3. "The lack of effective secure voice communications systems was identified as the most serious deficiency in Department of Defense communications."

The recommendations of the committee included reorganization of the DCA, improved quality of personnel assigned to DCA, and improvement in cost accounting and system management functions. Owing, in large part, to this Congressional interest, it can be expected that the hegemony of the DCA over the military departments will increase within the next decade. It is also reasonable to expect an increased interest in communications economy, both as a result of Congressional interest and the increasing competition for funds in government.

DATA TRAFFIC

The volume of data communications in the DCS will increase over the next 10 years, as it will in nearly all

other common-user communications systems on earth.[23] Primary causes of this anticipated growth in the DCS are:[24]

1. Increased volume of record communications.

2. Increased use of computers within the military services, with their concomitant requirements for intercommunications with remote terminal devices and other computers without human intervention. This will probably be the dominant growth category.

3. Increased emphasis on secure voice services.

The rates of growth in these categories will likely not be equal, and this may result in a wildly varying rate of digital traffic increase as one and then another of these factors dominates the scene in any given period.

It is important to define *data traffic* quite precisely, at this point, to preclude future confusion. Data traffic includes all record communications (except facsimile), the requirements for communications directly between digital machines, and digitized secure speech. Not included in this term are digital signals that may result from digitization of speech, as in a pulse code modulation (PCM) system. The reason for this particular distinction is to maintain comparability of statistics. Those items included under the heading data traffic have been recorded for some years, and the history of their growth is therefore known.

The same is true for voice traffic. If this distinction is not maintained, any comparison of traffic by categories in the past and in the future is invalid, as the requirements are driven by entirely different demands. Thus, when one examines the DCS traffic mix and imputes a certain value to data and voice traffic, it must not be implied that all of the voice traffic is carried by analog means; some may be digitized. Similarly, some of the data may be carried on analog trunks, as all of it is now.

The rate of growth of data traffic in the DCS is not well agreed upon. As important as determining a reasonably likely value for growth is the means of measuring the quantity of data traffic. Because the DCS is a speech-oriented system, its standard "building block" is the 4-kHz channel.[25] Thus, the only significant measure of data traffic or data requirements in the DCS is in terms of 4-kHz channel equivalents. This is not as much of a simplification as it may appear, because there is considerable disagreement as to the equivalency of a designated bandwidth and a rate of data transfer, but it is a beginning.

Projections of the growth of anything must be examined in the light of the capacity of the system to absorb the increase. The DCS is no exception. Although the determination of a "most likely" rate of increase of data traffic

in the DCS is beyond the limits of this paper, it is prudent to examine the extremes that have been proposed, to determine the capacity of the system to accept them.

Annual growth rates for data traffic of 25 to 30 per cent have been proposed for the DCS over the next decade; however, such rates tend to be self-limiting for several reasons. For example, a growth rate of 30 per cent annually would result in a doubling of the DCS data traffic in 2.3 years, which equates to the purchase of an additional increment to the DCS of approximately 20 per cent of the size of the present system within the same 2.3 years to keep pace with the increasing data traffic. On an annual basis, such an increase would cost approximately $200 million in operations and maintenance funds alone. Costs of that magnitude over such a short period are unlikely to receive Congressional approval, and thus alternative means of transmitting much of these data will have to be found.

On the other hand, it is difficult to argue for a rate of data increase much below 10 or 11 per cent, as just such an annual rate has been experienced since 1965. For these reasons an annual rate of growth of 11 per cent for DCS data traffic is used in this paper. It is to be understood that this figure refers to 4-kHz channel equivalents, not to numbers of data circuits or other often-used

denominators.

ANALOG SYSTEMS

Somewhat less obvious than the predictions above is the fact that the DCS of 1985 will utilize most of today's DCS equipment essentially as it presently exists. The reason for this seeming paradox is economic. Although the exact value of the installed DCS plant is not known, it is estimated to be between two and four billion dollars and increasing every year as new systems and links become operational.[26] Further complicating the picture is the extremely long life of fixed-plant telecommunications equipment. Although the government does not amortize this equipment cost as civilian industry does, it cannot afford to throw away hundreds of millions of equipment dollars without considerable military justification. The trade-off analysis that permits the obsolescence of a fighter aircraft to result in the development of a more technically advanced craft does not extend itself well into the field of military communications. It is harder to demonstrate either the relative advantage in combat of improved communications or the subjective value of the improvement. By way of comparison, the value of the DCS plant exceeds the value of the entire fleet of C5A aircraft[27] and will probably last much

longer. The Bell System still incorporates many equipments and links engineered in the 1930's for very similar economic reasons.

DIGITAL SYSTEMS

The DCS of the 1980's will contain some all-digital systems for two basic reasons. First, the Defense Satellite Communications System (DSCS), Phase II, which is scheduled to go into operation very soon, provides full capability for digital communications on both up- and down-link. Second, many of the existing DCS analog links are at or beyond their data-handling capacity. They must be replaced, and it would be economically and technically logical to make the replacement digital, particularly within a relatively small geographical area. This is not to imply that digital techniques would necessarily be introduced for voice traffic; more likely, data traffic would be segregated onto a newly installed digital system which would operate in parallel with pre-existing analog equipments for voice traffic.[28]

Digital transmission is likely in the DCS by 1985 on a link-by-link basis for a specific purpose. The demand for extremely long, high-quality links and expansion of secure voice service within small areas will most probably be responsible for the installation of such digital links.

TRAFFIC MIX

An annual rate of data traffic growth has previously been established at 11 per cent for use herein. Voice traffic during this period will not remain static either and may also be expected to increase, albeit somewhat more slowly than data. This is so because some of the new data requirements replace a previously engineered analog path for similar service and because the number of persons in the military, and their dispersion, is not increasing at the same rate as the use of computers and record communications.

A reasonable projection for voice traffic growth over the next decade appears to be in the vicinity of 2 per cent per annum.[29] Figure 1 indicates the traffic mix anticipated in the DCS over the next several years as a result of this rate, and the growth rate for the DCS overall. These relationships are not linear, and the DCS growth curve, especially, approaches an exponential rate of increase. This obviously cannot increase indefinitely without bounds, and the location of those bounds will require that some very difficult and far-reaching decisions be made in the near future.

As stated earlier, it cannot be assumed that the traffic mix shown in Figure 1 relates to transmission means. Voice traffic may be handled by digital means, and vice

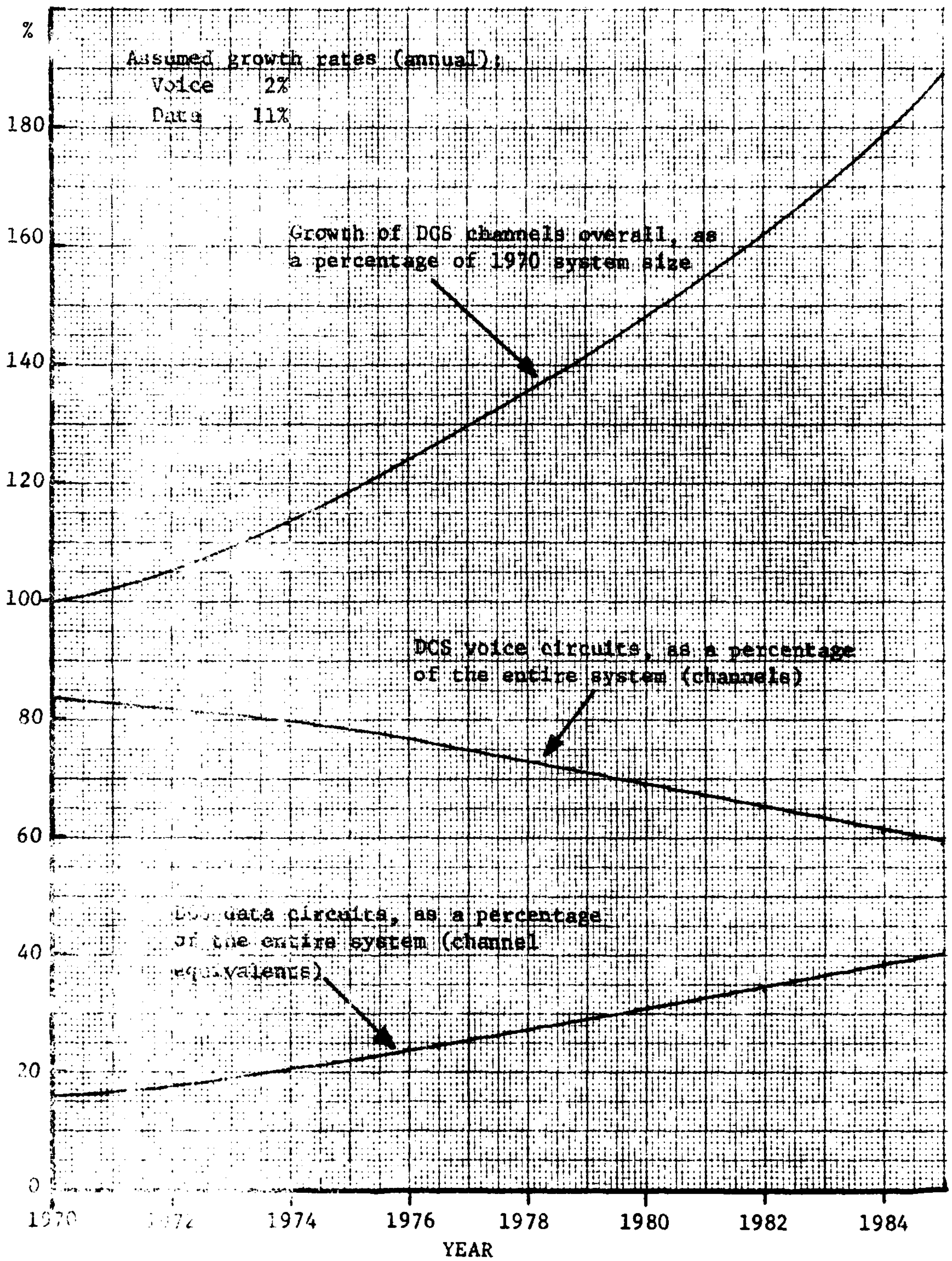


Fig 1--Defense Communications System Growth Trends

versa, according primarily to the economics of the situation. The traffic mix indicates traffic by requirement, not by transmission media.

These projections must be used with great care, for they are constructed from data which contain certain implicit assumptions; namely, data traffic will continue with the same efficiency that obtains today and an unlimited potential exists for data traffic growth. These assumptions are valid only under a very limited set of conditions. Unless one foresees the total elimination of interpersonal communications in favor of machines, there is obviously an upper bound on the growth of data traffic. This boundary will also depend upon the propensity of men to introduce further automated systems into use and the limits of the economy to produce and support such systems. Thus, data traffic cannot be thought of as ever-increasing in the long term. For the short term, however, if it is assumed that the network is not approaching data saturation, "unlimited" growth is an acceptable assumption.

A less acceptable assumption is that pertaining to the efficiency of data transmission. As handled today over analog channels, data transmission utilizes only a very small portion of the available capacity of the channel--in some cases less than 5 per cent and usually less than 15 per

cent.[30] There is a very large potential for improvement here, and an increase in the efficiency of data transmission effectively lowers the rate of growth of data transmission requirements. For example, if the channel efficiency were raised by a factor of two, the effective growth rate for data channel equivalents would be halved. Owing largely to economics, there is every reason to believe that more efficient transmission of data will be actively pursued in the next several years. However, the rate of growth assumed for data traffic is the lowest of those proposed by responsible analysts. Thus, although it is likely that the overall growth of DCS data traffic will exceed 11 per cent per annum in the next decade, this writer believes it will not significantly surpass this level after adjustment for increased transmission efficiency. Under these conditions, the traffic projection shown in Figure 1 is considered acceptable for planning and modeling purposes.

AUTODIN

The increase of record communications will impact heavily on AUTODIN. One can expect to see an increase in the capacity of existing switches and a provision for circuit switching overseas, as now exists on the Continental United States switches, to provide direct connection between

high-speed, high-volume subscribers to whom the delay of message switching is intolerable. Improvements in data transmission efficiency are a necessity in this network, and it is likely that such features as packet switching and Pierce loops will be incorporated into AUTODIN on a selective basis by 1985.[31]

AUTOVON

Of the major DCS subsystems, AUTOVON serves the most subscribers. It also has the most checkered performance record. Increased voice traffic requirements will require additional AUTOVON trunks and access lines. A likely place for initial installation of digital transmission facilities for speech would be on the long-haul AUTOVON trunks, as this would provide the greatest incremental signal improvement. If the AUTOVON can provide service comparable to that of the Direct Distance Dial network of the Bell System, it is reasonable to suppose that increased use of it will be made for data transmission by casual users equipped with acoustic couplers or similar devices. This could be disastrous unless closely controlled, because AUTOVON is not designed to carry a large volume of data traffic.[32]

AUTOSEVOCOM

The AUTOSEVOCOM network will probably show the

smallest change over the next decade, owing to the considerable cost of adding subscribers and the fact that the most important subscribers are already connected to the network. A further inhibiting factor in the growth of this subsystem is the likelihood of digitized voice being incorporated into the DCS on a large scale, which would effectively eliminate the justification for AUTOSEVOCOM as a distinct subsystem.

A DCS MODEL

To evaluate the effect of expected developments or phenomena on the DCS of 1985, it is necessary to create a model of that system because the reality does not yet exist. The model must be simple enough to permit reasonable evaluation, yet complex enough to accurately portray the effects of changes upon the DCS by extrapolation.

It is possible to construct a mathematical model of the 1985 DCS which encompasses most, if not all, of the variables of interest. Such attempts have been made previously on other systems and have proved so complex that only trivial analyses were performed with them.[33] It can be argued that such a model is required to account for the synergistic effect of multiple system parameter changes; however, experience has shown that superior results are achieved in practice from a comprehensible model that

considers one variable at a time.[34] This approach is especially attractive in this case, as many of the parameters to be considered defy precise quantization. Figure 2 presents the topography of a model that is utilized in this study to simulate the DCS backbone.

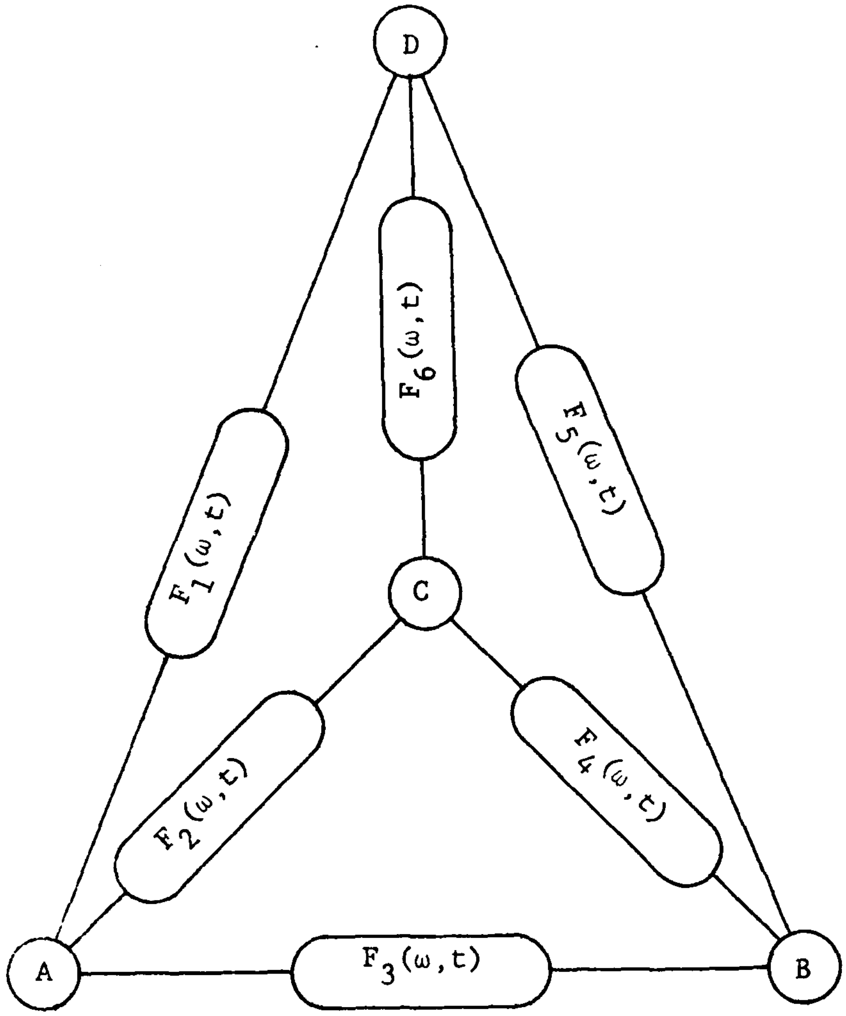
The model in Figure 2 is not a closed system; other links and users can be connected to any node. However, the model as shown is felt to be adequate to describe the performance of the DCS under the influence of the variables considered herein. Technical characteristics of importance in the model (as at 1985) are:

1. The maximum circuit length built up will be six links of three sections each, for a total built-up length of 12,000 nautical miles, or twice that now accepted in the DCS. (A section does not involve multiplexing/demultiplexing at connection points.) The overall signal-to-noise ratio (SNR) for speech will be 29 db or better.[35]

2. The traffic mix will be approximately 40 per cent data and 60 per cent voice in terms of 4-hHz channel equivalents (see Fig. 1, page 21).

3. AUTOSEVOCOM will not be considered as a separate subsystem but as a portion of the data traffic load.

4. AUTODIN will account for approximately 5 per cent of the dedicated circuits, but data traffic will no



Y : Nodal point interconnecting system links and permitting user access and switching. Multiplexing performed at the node.

$F_n(\omega,t)$ Channel, whose transfer function is dependent upon both frequency and time. Includes repeaters and regenerators, but not multiplexers.

Fig. 2.--Defense Communications System Model

longer be restricted to full-time private user routes.[36] The increase in data traffic will be reflected in part by increased AUTODIN transmission efficiency.

Examination of the model reveals that it is capable of simulating circuit restoral and traffic delivery by means of alternate routing. The characteristics of transmission media are accounted for by the proper definition of the channel transfer function, $F_n(\omega,t)$.

CHAPTER III

ANALOG AND DIGITAL TRANSMISSION CONCEPTS

BACKGROUND: ANALOG TRANSMISSION

Although the development of operating electrical telegraph instruments occurred several decades prior to the development of the telephone, the latter initiated the "communications revolution" of the late nineteenth century and is still today the primary driving force behind development of communications networks. The result has been that throughout the world the primary communications networks are based upon telephonic communications, with record communications being adapted to fit into that system.

The near-instant popularity of the telephone is attributable to two primary causes: the state of technology at the time of its development and soon thereafter and man's preference for speaking with other men rather than writing them messages. The first commercial telephone exchange opened in 1878, and within 40 years technology had progressed to the point where frequency-division multiplex was already in use.[1] All-metallic circuits had proliferated long before that time, and the telephone was well-estab-

lished as the dominant means of communications in the United States. In that same 40-year period, wireless telephony became practical and the further development of the vacuum tube encouraged increasing use of multiplexing and transmission of analog telephone signals.

Man's propensity to conduct his business orally has not significantly changed in the past century, either, for the Bell Telephone System has shown an annual growth rate of approximately 19 per cent in voiceband channels since 1959.[2] In 1966, more than 800,000 toll trunks alone existed in that network,[3] and the number today is likely well over one million.

Such is the magnitude of the telephone network that scarcely a single person in the civilized world is unfamiliar with a telephone instrument. The telephone systems of the world, although not entirely compatible, are interconnectable, with the result that no point in the world is very distant from another in the communications sense. These telephone systems, owing to their size and omnipresence, have become the primary route for virtually all common-user communications in the world, to include record traffic. Telephone systems are primarily analog systems, however, and thus the non-telephonic signals impressed upon them must first be adapted for transmission by this means.

In this chapter the nature of both analog and digital signals is reviewed. The definition of a signal as digital or analog depends upon its form for transmission, not on the nature of the source signal. Thus, a teletype signal, for example, is categorized as an analog signal if it has been so converted for transmission.

ANALOG SIGNAL ANALYSIS AND MODULATION

The most basic analog signal being the sine wave, it is understood that in the general case[4]

$$E(t) = A \sin (\omega t + \phi)$$

where: $E(t)$ = the function under consideration

A = sine wave amplitude

ω = natural (radian) frequency

t = time

ϕ = the phase angle.

Briefly reviewing modulation theory, a pure sine wave, being predictable at every point in time, conveys no information but can become an information carrier in the event one of the factors A, ω, or ϕ is made a time-varying information function. In order, this leads to amplitude, frequency, and phase modulation, each of which is considered separately below. Further, modulation of a sinusoidal carrier produces both a carrier frequency component and

information-bearing sidebands, the location of which, relative to the carrier frequency, is dependent upon the modulating frequencies and the waveform of the modulating signal.[5]

Figure 3 summarizes the frequency spectrum of a sine wave amplitude modulated by a square wave. No information is conveyed by the carrier component, and the information of the upper and lower sidebands is identical. Recognizing this fact, one quickly sees that transmission of more of the signal than is required to convey the necessary information is inefficient. This realization has led to the development of several variants of amplitude modulation (AM), notably double-sideband AM (DSB), in which both sidebands are transmitted either with or without the carrier component; single-sideband suppressed carrier (SSSC or, more commonly, SSB), in which only one of the sidebands is transmitted and the carrier component is greatly suppressed; and vestigal sideband (VSB) transmission, in which one sideband is greatly reduced in amplitude prior to transmission. It can readily be seen that SSB, compared with DSB, requires less bandwidth for transmission. Given the same transmitter peak power, SSB enjoys an advantage of approximately 9 db over DSB in signal-to-noise ratio (SNR).[6] For this reason, SSB is used in preference to other AM methods in virtually all AM

frequency division multiplex (FDM) systems.

It can be shown that frequency and phase modulation may be considered together since they are different sorts of angle modulation.[7] Figure 4 shows the frequency spectrum of a square wave modulating a carrier in FM with a deviation ratio of unity. The amplitude of the higher-order components decreases more rapidly than was the case with AM. For integer deviation index values, the FM signal consists of two square-wave AM signals centered on the Mark and Space steady-state frequencies and keyed in opposite phase.[8]

As compared to DSB-AM, FM has a theoretical noise advantage of approximately 15 db,[9] which is obtained by trading bandwidth for SNR as well as by the nature of the FM signal. It is also approximately 12 db less sensitive to impulse noise than AM systems,[10] a considerable advantage in data transmission.

Although its frequency spectrum is virtually the same as that of FM, phase modulation (PM) has a theoretical noise advantage of 2 db over FM and is less sensitive to noise interference for the same bandwidth.[11] Unhappily, practical PM systems have proved extremely complex in the past, and this has eradicated much of the theoretical noise advantage. Recently, however, developments in integrated circuits have made complex circuitry much less costly, and

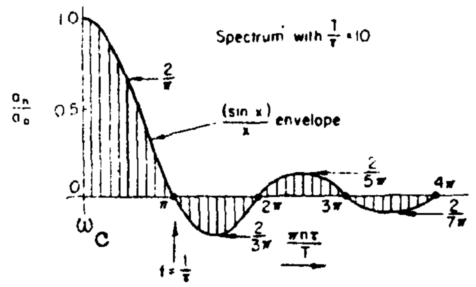


Fig. 3.--Fourier series representation of upper sideband of a sine wave carrier of frequency ω_c modulated by a rectangular pulse train with a pulse spacing ten times the pulse width. (Adapted from Bennett & Davey, p. 34.)

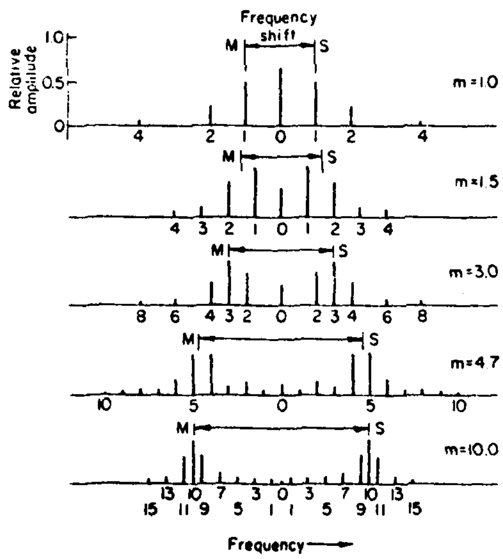


Fig. 4.--Square-wave FM spectra, deviation ratio (m) from 1 to 10. (Bennett & Davey, p. 39.)

PM is becoming much more common as a means of data transmission.

Because both FM and PM occupy more bandwidth than an SSB signal derived from the same modulating wave, they have not been utilized as widely as SSB in FDM systems in spite of their noise advantage. However, the transmission of data over telephone systems requires that the data signal be converted into analog form prior to its insertion into the system, and this conversion is commonly performed utilizing either FM or PM, largely owing to the performance of these systems in noise.

TRAFFIC FLOW IN ANALOG SYSTEMS

The telephone is an instrument intended to permit two or more persons separated by distance to converse. By its very nature, conversation is interactive and adaptive and requires real-time communication between the concerned parties. In contrast, the instantaneous content of a broadcast radio message is not subject to real-time interaction by the listener. The telephone network has been developed to permit interactive communications.

A conversation, unlike a written treatise, does not lend itself to storage for retransmission at a later time. One-half of a conversation is meaningless without the other

half, both at the same time. For this reason, telephones must be interconnected in real time, which requires tandem connection of telephone circuits. Since 1878, the development of circuit switching in telephone networks has proceeded to become a science well-defined mathematically, and much of the capital investment of any telephone network is in circuit switching equipment.

Telephone systems are designed with the application of statistics to provide a given grade of service at busy hour. This is usually expressed in percentage of calls refused on a given path. (No telephone system is designed to permit all subscribers to place calls at any one time; this is economically infeasible. Rather, a certain refusal rate is accepted from the start to permit design of a usable, affordable system.) If the stated grade of service can be achieved during busy hour, however, it is obvious that a higher grade of service is achievable off-busy hour, as the circuits are still there that were there to handle the busy hour load. Owing to the evanescent nature of conversation, however, this capacity is wasted unless the demand for service can be spread more evenly through the operating day. This is the driving force behind lowered telephone tariffs during off-peak periods and advertisements that encourage people to call relatives during the evening

hours rather than during the busy daylight hours. Stated another way, because a telephone network requires circuit switching as a routing philosophy in order to function as it must, it is necessary to over-build the network so as to provide the required service quality at the busiest time of the day. This fact has important connotations for the addition of data communications to the network.

The basic building block of the telephone system is the voice channel, which has a bandwidth that extends from approximately 300 to 3400 Hz.[12] (This is referred to as a 4-kHz channel rather than a 3-kHz channel because the addition of guard bands in an FDM system causes the spectrum occupied by a voice channel in the multiplex system to be approximately 4-kHz.) Channels are engineered between points based upon busy-hour requirements and, once installed, remain in place until physically disconnected. Although it is possible to reroute circuits in response to traffic demands through use of technical control facilities (automatic or manual), this technique is not in wide use, with the result that the point-to-point capacity of the telephone system, measured in bandwidth, is largely fixed and unresponsive to short-term variations in traffic flow.

SIGNAL STATISTICS

A communications system must be designed with respect not only to the traffic flow requirements imposed upon it but also the content of the signals it carries. The primary signal carried by the telephone network is that of the human voice, which deserves description.

Considerable statistical analysis has been directed toward the human voice, and the following several things have been determined about it in the telephone system context:[13]

1. Voice levels at the input to a channel modem are typically in the vicinity of -16 dbm.

2. Although the dynamic range of voice signals can be in the range of 70 db, there is only a 1 per cent chance that a voice signal will exceed -3 dbm at a given time.

3. The probability that a channel will be active at any given moment during busy hour does not exceed 0.25.

4. Human speech is informationally redundant and has a very low information efficiency, as defined by Shannon's criteria.[14] In fact, the information content of speech is so low that only about 25 per cent of a voice signal is required for intelligibility.[15]

5. Human speech is extremely random in amplitude and phase and, as a result, possesses a low average power

level.[16]

The characteristics of human speech have driven the design of telephone networks. This must be compensated for when applying signals other than voice to the network, for non-voice signals do not share the same statistical qualities as speech, and the effect of this on the network can be significant, especially as regards average channel power.

SIGNALLING

A system as complex as the telephone network requires a considerable amount of supervisory communication, which is accomplished by the use of various signals to cause the system to perform the desired function on command. In the analog case, these signals consist of tones, singly or in combination, which are transmitted with the information signal either within the information channel (in-band signalling) or in a separate supervisory channel (out-of-band signalling).[17] In the case of the former, the supervisory signals are removed from the information signal before it is routed to the ultimate subscriber.

Signalling in a digital telephone system is accomplished by digital signals that contain routing and supervisory data, much as in a computer communications network. This type of signalling has not been widely employed in

telephone systems but will be necessary in a digitized system to avoid repeated analog-to-digital conversions. The inherent differences between analog and digital signalling necessitate extensive interfacing when links utilizing different modes are connected in tandem.

When in-band signalling is used on a telephone channel, it frequently occurs that the signal tones interfere with the data signal introduced on the channel and cause transmission errors. The same situation can occur with crosstalked signalling tones from adjacent channels. The usual solution to this problem is to disable the signalling equipment when the circuit is used for data. An alternative solution is the use of out-of-band signalling to preclude this problem.

SWITCHED TELEPHONE NETWORK TRANSMISSION IMPAIRMENTS

A major portion of communications system design concerns the minimization of signal degradation caused by the imperfections of the channel, as no communications channel is transparent. The telephone networks of the world have been optimized for transmission of voice signals which are particularly vulnerable to noise, distortion, crosstalk, echo, FDM frequency instability, and level variations.[18]

Of the two forms of noise, impulse and white, the

human ear is relatively insensitive to the former, with the result that little effort has been expended in controlling it in telephone systems until quite recently, and then in deference to increased data traffic in the system. White noise, on the other hand, strongly affects the intelligibility of speech, with an occurrence of 10 db SNR generally considered the minimum acceptable for communications. The design target minimum is usually much higher, on the order of 17 to 23 db.[19] Considerable effort has been spent in minimizing the effects of white noise on voice signals, the more common methods being the use of the strongest possible channel signals (consistent with adjacent channel interference criteria and system loading constraints), SSB modulation, and compandors. That these efforts have been generally successful is reflected in the infrequency with which a noise-limited built-up connection is encountered. A recent study of the Bell System indicates that the average noise on toll trunks (C-message weighting, for all types of carriers) was 18.4 dbrnc,[20] which is a quiet circuit indeed.

Distortion occurs in three primary categories: amplitude, intermodulation, and phase, with the human ear being quite insensitive to phase distortion.[21] Thus, as with impulse noise, no real effort to control it was made for reasons of speech intelligibility. Intermodulation

distortion in multiplex systems is heard as a form of noise on the signal; however, it is undesirable for a number of non-speech intelligibility-related reasons and has been well controlled by amplifier design and close attention to multiplex system loading.

Amplitude distortion (the uneven response of a channel over a band of frequencies) is somewhat troublesome to the ear, although the problem it causes is generally not one of intelligibility but, rather, one of fidelity.[22] Nevertheless, it has been minimized in most telephone systems through the use of loaded cables and, more recently, active and passive equalizers. The amplitude distortion characteristics of several types of Bell System equipment are shown in Figure 5 for illustration.[23]

Crosstalk has a deleterious effect on speech intelligibility, especially during the quiet portions of a conversation, at which time it is much more noticeable than during active speech. Crosstalk has been a problem since before the use of amplifiers, and methods of reducing its effects have become quite well refined. These methods generally involve the judicious selection of signal paths in voice-frequency and carrier-frequency multipair cables, using such methods as frequency-staggering or random splicing, or the use of compandors. Compandors offer a

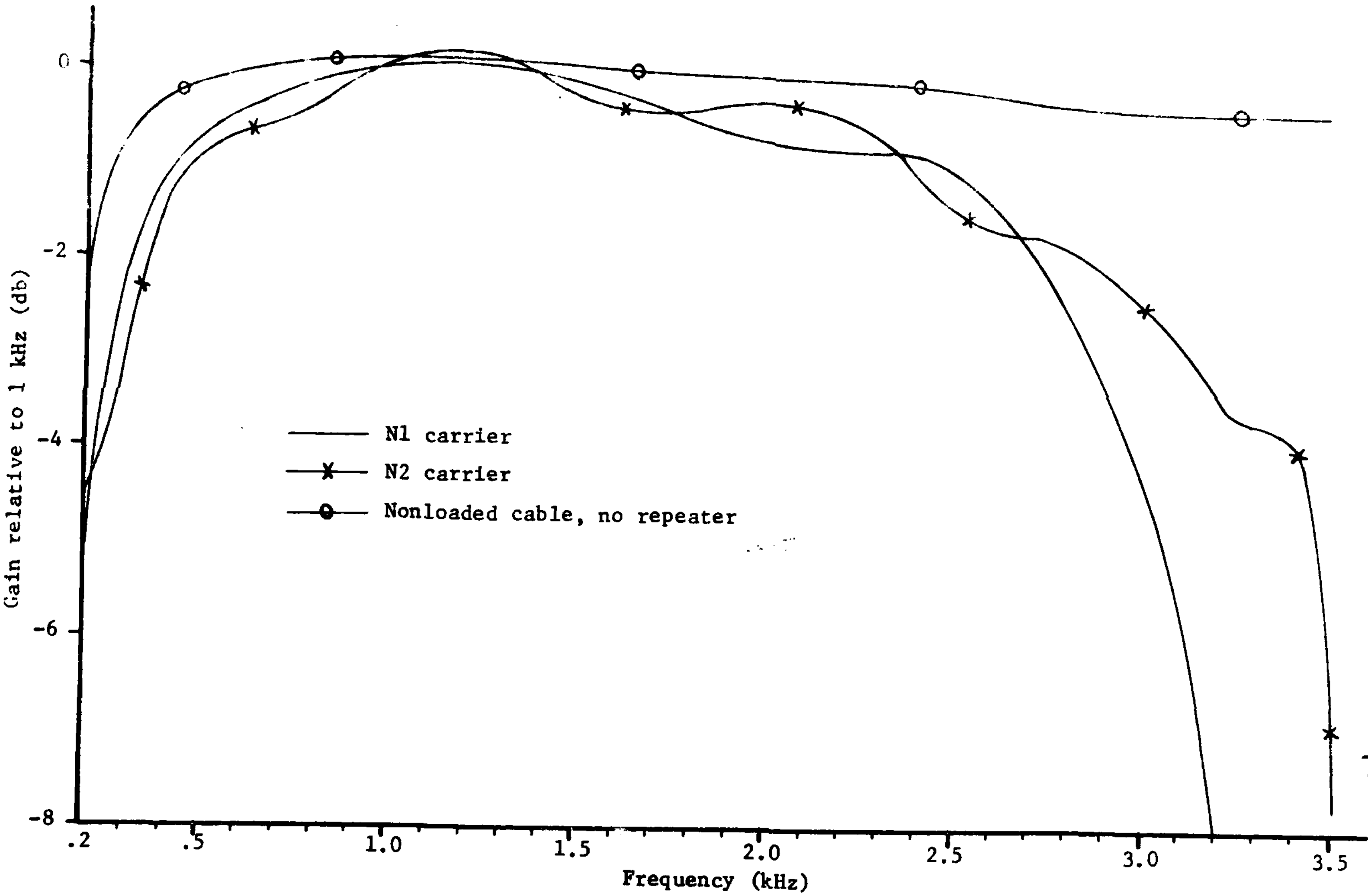


Fig. 5.--Amplitude Distortion Characteristics of Carrier Systems and Cable

20 to 28 db reduction in noise and crosstalk on the typical circuit and are widely used.[24]

Echo, also a troublesome transmission impairment, has worsened since the advent of the communications satellite. The primary source of telephone echo is the hybrid transformer used to convert 2-wire loop plant connections to 4-wire toll plant, and an obvious solution to the problem is to eliminate the hybrid. This has been done with good results in some specialized systems, but the capital investment in 2-wire loop plant makes such a solution economically infeasible in the general case. Speaker tolerance to echo, which is very dependent upon both delay time and loudness, varies from person to person. It is generally considered that echo delays exceeding 45 msec must be corrected by suppression techniques. An echo suppressor, in its most basic sense, senses the direction of conversation and inserts a large loss in the opposite path to attenuate the echo. Considerable refinement of this technique has been undertaken to permit 2-way simultaneous conversation and to alleviate other shortcomings, but the basic approach remains the same and is considerably less than perfect.[25]

For tandem-connected FDM systems to operate properly, it is necessary that they exhibit a high degree of frequency stability so that successive modulation-demodula-

tion steps will not cause excessive translation of the signal in the frequency domain. Frequency shifts of ±15 Hz are considered acceptable for human speech, but technical requirements generally impose a much smaller tolerance upon the system, to the extent that frequency stability in FDM systems is seldom a problem of concern to the user.[26]

A similar situation obtains with long-time constant level variations which affect the channel level envelope. Such variations are extremely annoying to the persons speaking over the channel, but they are also relatively simple to compensate through the use of variable-gain amplifiers whose transfer function is dynamically controlled by a reference signal such as a transmitted pilot tone.

The impairments discussed above prove most troublesome to voice communications and thus have received the most attention in the design of telephone networks over the past half century, and longer. A very considerable portion of any telephone system's capital investment lies in equipment designed to optimize the network for the transmission of voice, and this fact results in an inertia factor which is encountered whenever the technical modification of the network is required, for whatever reason. It must be borne in mind that the investment represented by any telephone system of major proportions, particularly the DCS, Bell

System, and most foreign PTT organizations, is substantial, and it is economically impossible to replace the entire plant in a short period of time. The telephone system engineer's largest single constraint is that he must utilize the equipment in place for whatever new service he is presently designing, and he must acknowledge that the equipment presently in service has an economic life of 40 to 50 years in most circumstances. This is true even in the DCS, although the life of equipment may be taken as 20 or 30 years. As digital transmission is discussed, this effect of investment upon drastic and sudden technological change must be remembered.

DIGITAL COMMUNICATIONS HISTORY

Although often obscured by press agentry proclaiming a "computer revolution," it is interesting to note that digital electrical communications antedates analog transmission by more than 30 years. However, many of the techniques of digital transmission changed very little for a century, as digital circuitry became economically and technically feasible on a large scale only with the development of the transistor in 1948. Today's rapid proliferation of digital equipment of all sorts is a direct result of the perfection of integrated electronics and large-scale integration (LSI)

which have made possible processes only theoretically described in previous years.[27]

Until the mid-1950's digital transmission took the form of telegraphy in nearly all instances, with communications speeds of 100 to 300 words per minute (WPM) the maximum obtainable owing to the electromechanical nature of the equipment involved. Even the first generation of computers in the early 1950's did little to change this state of affairs, for those machines were used to automate functions that were too unwieldy for human effort. As computers became more commonplace in the late 1950's, however, the need to intercommunicate manifested itself, and digital techniques previously applied only to the internal design of computing machines were applied to the communications interfaces between these machines.[28] Only one communications system existed that reached nearly every point of interest on the map: the telephone network. Telephone circuits rapidly became carriers of much more than speech, and this continues today. A brief review of digital modulation theory follows, and the implications of this on communications systems are then discussed.

DIGITAL SIGNALS AND MODULATION

In contrast to an analog signal, which can assume

any value of amplitude within the specified range, a digital signal is constrained to certain predefined discrete levels within a specified range. The transitions between levels, or states, occur instantaneously in theory, although not so in fact. Nevertheless, the transitions are extremely abrupt, with the result that the mathematical description of a digital signal is a multiform function, or a series of discontinuous functions which are considerably less convenient to analyze than the uniform transcendental functions that describe analog signals.

A mathematical development of the nature of digital signals is beyond the scope of this discussion, but there are several works on that subject.[29] For the purpose of review, the "A" portion of Figure 6 shows the Fourier series spectral envelope for a rectangular pulse. The low-frequency components of the Fourier series are responsible for the flatness of the top of the pulses, and the high-frequency components are those that determine the steepness of the rising and falling edges of the pulses. For comparison, the "B" portion of Figure 6 shows a cosine pulse spectrum to demonstrate the diminished high-frequency component amplitudes due to the increased rise and fall times of this pulse versus the square pulse.

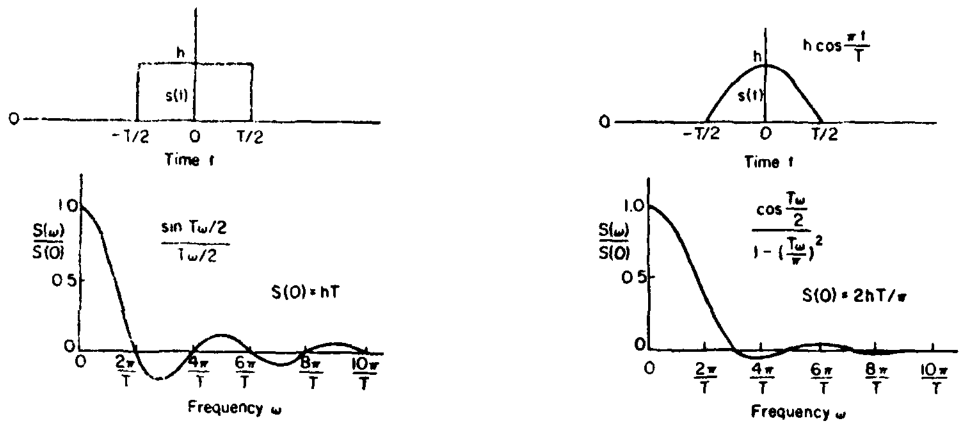
Digital signals can be used to modulate a sinusoidal

carrier in the same manner as can analog signals. Amplitude modulation of a carrier by pulses amounts to on-off switching of the carrier at the modulating rate. This is shown in Figure 7 and is exactly analogous to continuous wave (CW) telegraphy. As with any other form of AM, this system is especially vulnerable to noise and therefore is presently in only limited use. Its primary advantage is the simplicity with which it may be implemented.[30]

More commonly, digital signals are used to frequency or phase modulate a sinusoidal carrier. These cases are also shown in Figure 7. Both FM and PM, as discussed previously, enjoy a substantial noise advantage over AM, and this is one of the primary reasons for their wide use with data signals. When used with a binary data signal, FM modulation is often referred to as frequency-shift keying (FSK), which has been in use on radio and carrier links for many years for the transmission of teletype data.

The use of analog techniques for the transmission of data, as described above, has been the method of choice for many years. However, the advent of practical and economically feasible digital circuitry has made the use of digital carriers quite attractive.

A digital carrier is exactly equivalent to the analog case. It is a train of pulses of a constant



A B

Fig. 6.--Spectral amplitude functions: A, rectangular pulse; B, cosine pulse. (Bennett & Davey, pp. 50-51.)

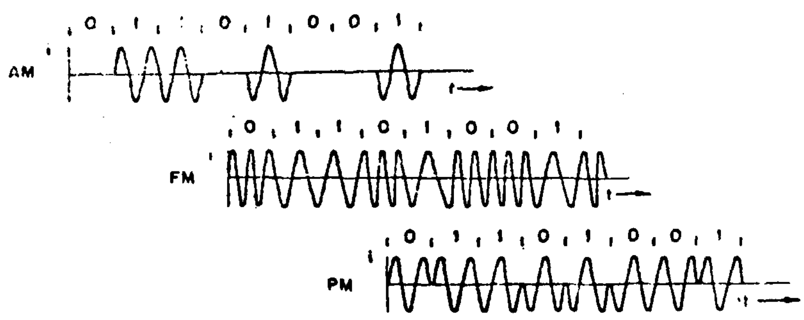


Fig. 7.--Digital modulation of a sinusoidal carrier. (Bennett & Davey, pp. 29-30.)

frequency (repetition rate) and amplitude, whereas the analog carrier is a sine wave of constant frequency and amplitude. As might be expected, a digital carrier may be modulated by varying its amplitude, frequency, or phase. These forms of modulation are known respectively as pulse amplitude modulation (PAM), pulse duration modulation (PDM), and pulse position modulation (PPM) and are depicted in Figure 8.[31] It is worthy of note that the modulating signal may be either analog or digital in form, as in the analog case.

All of these digital modulation schemes have been utilized in communications equipment at one time or another; however, each suffers from the same shortcomings as its analog counterpart. The primary enemy of these systems is noise, as in the analog systems, for it creates uncertainty in the location of the pulse top, edges, or location in time, and therefore results in a loss of signal fidelity just as previously discussed. To exacerbate matters, these types of modulated carriers are more sensitive to certain transmission impairments, such as phase delay, than speech signals. This requires special conditioning of those telephone circuits used for their transmission.

The development of information and coding theory by Claude Shannon,[32] however, resulted in the development of a

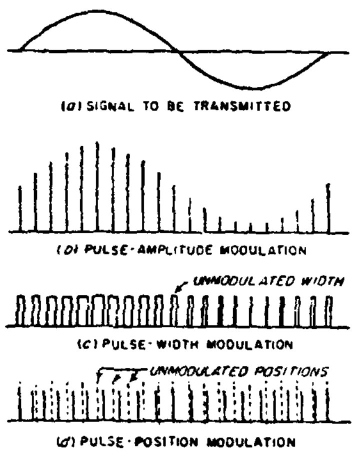


Fig. 8.--Types of Pulse Modulation
(Terman, p. 967.)

modulation technique which is superior to any of those already discussed, particularly with respect to its noise tolerance. This technique is somewhat erroneously called pulse code modulation (PCM). It is not so much a modulation scheme as a coding technique, but this is a point of philosophical difference. To date the primary use of PCM has been in the transmission of speech.

PCM is a system whereby a signal is analyzed by a predefined set of rules, and its value at certain moments is sampled. A number is assigned that value, and a digital code representing that number is transmitted. At the receiving end, the code is used to activate a generator that produces an output signal of the amplitude originally sensed by the transmitting coder. These signals are assembled sequentially to produce a replica of the source signal.[33] The success of PCM rests upon sampling theory, coding theory, and the technology required to perform the complex electronic functions associated with the above sequence of events.

Nyquist[34] has shown that a suitable replica of any given signal can be reproduced from a number of samples taken of the source signal. In simplest form, Nyquist stated that, to assure fidelity in sampling, the source signal must be sampled at a rate that is at least twice that

of the highest-frequency component in the source signal. Since the highest frequency of human speech usually accepted by a telephone system is approximately 3400 Hz, a voice signal must be sampled at a rate of at least 2 x 3400, which means 6800 Hz. In practice, a rate of 8000 Hz is generally used.[35]

If a number is to represent the sampled value, it is necessary to establish a certain number of permissible levels that the signal can attain. The analog signal, not knowing the constraints, will continue to assume whatever value it chooses, but a detector will respond to any signal value within the limits of a predefined range by assigning it the number associated with that range. The quality of the reconstructed signal depends upon both the frequency with which the source signal as sampled and the number of levels into which it is quantized. As previously noted, the frequencies contained in the source signal are normally the limiting factor on sampling rate, and the bandwidth available to transmit the coded signal data (the PCM signal) is the limiting factor on the number of quantizing levels. In modern telephone practice 128 levels are usually used to encode human speech.[36] Figure 9 illustrates the methodology behind PCM.

The primary advantage of PCM is that the signal

[1] Pulse code modulation. Encoding for two waveforms is shown here. For each quantization level there is a corresponding digital code.

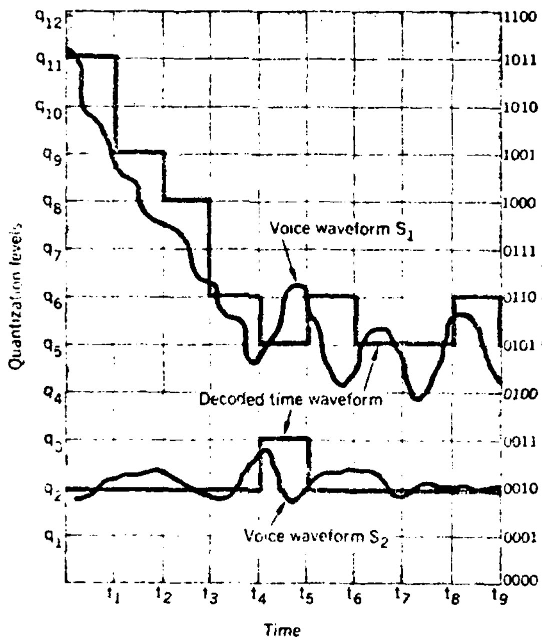


Fig. 9.--Pulse Code Modulation
(Bayless, Campanella, & Goldberg, p. 28.)

itself is not transmitted but, rather, is coded information that permits a reproduction of the signal. The effect of transmission impairments on the PCM signal is insignificant so long as the data can be recognized at the receiving end as the proper quantizing level code. Thus, techniques such as pulse regeneration and error-detecting coding may be used to enhance the durability of the PCM signal as it passes through the communications system. In theory, a signal may traverse a very noisy channel and be reconstructed with virtually no degradation as compared with the original.[37] In practice, performance very nearly as good is attainable.

It is apparent from an examination of the encoding process that PCM, owing primarily to the error in precision associated with the quantizing process, is incapable of producing a perfect replica of the source signal at the receiver. Any signal level within the bounds of a given level will be encoded as that level and will be duplicated at the receiver as that level rather than the precise level of the source. There being a practical upper bound on the number of quantizing levels, one can see that there will always be a certain amount of this error.[38] The goal of the PCM process is to minimize encoding error, which is called quantizing noise because it manifests itself as a random, noise-like signal superimposed upon the information signal.

The primary means of minimizing quantizing noise are by increasing the number of quantizing levels and by the use of compandors prior to quantization. The latter course of action is preferred, and it permits the use of non-uniform quantizing levels. Levels that occur frequently are encoded quite precisely; those that occur infrequently are encoded coarsely.[39]

The equipment required to realize good quality PCM is extremely complex, and until recently it was not economically competitive with FDM carrier equipment.[40] Developments in solid-state technology in the last decade, however, have altered the relative economic advantage of the two types of systems, and PCM is now widely used in both commercial systems (Bell T1 and T2 carriers) and military systems (e.g., the AN/TRC-111).

In an effort to improve upon the basic concept of PCM, several variants have been developed. Virtually all of these systems use some form of level predictor to attempt to lower the PCM data rate by taking advantage of the redundancy inherent in human speech.[41] The most promising of these techniques is delta modulation, which enjoys the advantages of simplicity and low cost as well as good signal quality. Delta modulation is in commercial use in Europe and is being studied carefully for use in the United

States.[42]

Still another method of transmitting speech by digital means is by use of a vocoder. Vocoders perform a spectral analysis of the source signal and transmit signals which cause the production of a like spectral sound at the receiver to reconstruct the source signal.[43] These devices are exceptionally complex, and, as anyone who has utilized a secure voice system recently will attest, they are not noted for good speech quality. Considering the advances being made in the field of PCM and its derivatives, it is likely that vocoders will fall into disuse in favor of PCM techniques unless a technical or economic breakthrough alters the relative advantages of the two systems.

Much of the above discussion concerns the digitization of speech for transmission on a digital system. The output of a PCM system or vocoder may be used as the modulating signal for either an analog or a digital carrier, and both schemes are in use.[44] Signals which originate in digital form are not usually subjected to further encoding prior to modulation, for they already possess the positive attributes discussed above for PCM pulses. The common exception to the above is when encryption is required.

TRAFFIC FLOW IN DIGITAL SYSTEMS

Until recently, digital traffic was synonymous with record traffic, for only telegraph messages were transmitted in digital form. As a result, the routing philosophy associated with digital traffic was one of non-real-time message switching, the epitome of which is the "torn-tape" teletype relay center.

Because different precedences are assigned to record traffic, each associated with a maximum acceptable system transit time, it is possible to more evenly distribute the traffic load in a record network. This is quite simply done by sending messages in order of urgency, with the routine ones being sent in off-peak periods. This practice was developed quite early in the design of teletypewriter networks and continues today in the AUTODIN system,[45] which is nothing more than an automated tape relay system.

When digital techniques are used for the transmission of speech, however, the same criteria concerning real-time interaction as discussed for analog systems come into play. For this reason one cannot generalize as to the routing philosophy that "should" be used in a digital communications system. It happens that most existing digital networks are store-and-forward networks, but this is an accident of history, not a technical constraint. The

optimum means of routing traffic in an all-digital system may well be circuit switching. On the other hand, it may be a combination of circuit and message switching, and this can only be determined by detailed study of the problem.

DIGITAL SIGNAL STATISTICS

In contrast to voice signals, it can be shown that data signals exhibit a much higher average power.[46] This is largely due to the higher information content of data signals but also has to do with the levels at which they are commonly applied to carrier systems. For this reason, when standard telephone systems are used for the transmission of data, considerable care must be taken to avoid power overloading of the multiplex system.

The activity of digital channels carrying high-speed traffic is frequently extremely high and often approaches 100 per cent. Also, digital transmission is very often full-duplex, as contrasted to the half-duplex nature of voice communication.[47] This can also pose serious power loading problems on multiplex systems unless considered and engineered for.

Low speed and batch-processing terminals, on the other hand (such as time-shared computer terminals), exhibit a channel activity factor of the same order as voice

signals. For a variety of reasons that are primarily economic, this low activity characteristic is being eliminated to permit greater utilization of connecting circuits, with the result that there is little practical difference in the activity factor to be associated with high-speed, high-volume circuits and these.[48]

DIGITAL RESPONSE TO
TRANSMISSION IMPAIRMENTS

Whether the signal transiting a communications system is digital or analog in form, it is assaulted by the same array of transmission impairments. However, the response or sensitivity of analog and digital signals to the same impairment differ as discussed below.

Generally speaking, digital signals are less sensitive to white noise than are analog signals,[49] and some very sophisticated techniques have been developed for the detection of digital signals in the presence of noise. In communications systems, the techniques used are usually variants of pulse regeneration. That is, at specified intervals the digital pulse train is regenerated by pulse slicing and is retransmitted as a "clean" replica of the noisy input.[50] This capability to regenerate pulse streams accounts for much of the noise advantage of PCM over FDM in practice.[51]

Impulse noise, on the other hand, is a significant

problem for digital transmission, for a noise impulse can masquerade as a signal pulse, which results in a spurious response at the receiver. The effect of impulse noise is countered by efforts to minimize the noise itself and by the use of error-detecting and/or correcting coding on the pulse stream.[52] Both these approaches are effective, but, when the suppression is implemented without regard to the magnitude of the impulse noise problem, both can be expensive. Complicating the suppression problem is the fact that a major portion of the impulse noise in telephone networks originates in electromechanical central office switchgear[53] which cannot be isolated from the system.

To faithfully reconstruct a data waveform after transmission, it is necessary to recover all components of the Fourier spectrum in the original amplitude and phase relationship.[54] Deviations from this phase and amplitude relationship result in distortion of the pulse, the seriousness of which is primarily dependent upon which components of the signal are affected by distortion. Both amplitude and phase distortion, which increase as the signalling speed is increased, are therefore of concern in data communications.[55] It is possible to correct both phase and amplitude distortion in a single equalizer, and this technique is often used in data transmission. As previously noted, the

ear is relatively insensitive to phase distortion, and thus this equalization is not noticeable when the same circuit is used on a time-shared basis for both speech and data.

Increasingly, data communications take place between different points at different times on a dial-up basis as opposed to the previous norm, which provided dedicated communications over a given path. Since every circuit exhibits different phase and amplitude characteristics and even between the same end points there is little likelihood of interconnecting the same sequence of circuits in repetitive tries, the problem of equalization can become serious. Proper equalization of a circuit is time-consuming and requires expert technical adjustments at both ends. As this is quite unsuitable for a dial-up system, considerable effort has been expended on the development of adaptive automatic equalizers that will analyze the circuit and equalize it in a matter of a few milliseconds. Start-up (adaptation) times of 50 msec are currently available in commercial adaptive equalizers for data use over the switched telephone network.[56] For illustrative purposes, typical phase distortion curves of common telephone carrier systems are shown in Figure 10.[57]

Crosstalk is also a problem with data transmission, for the crosstalking pulses can be easily mistaken for the

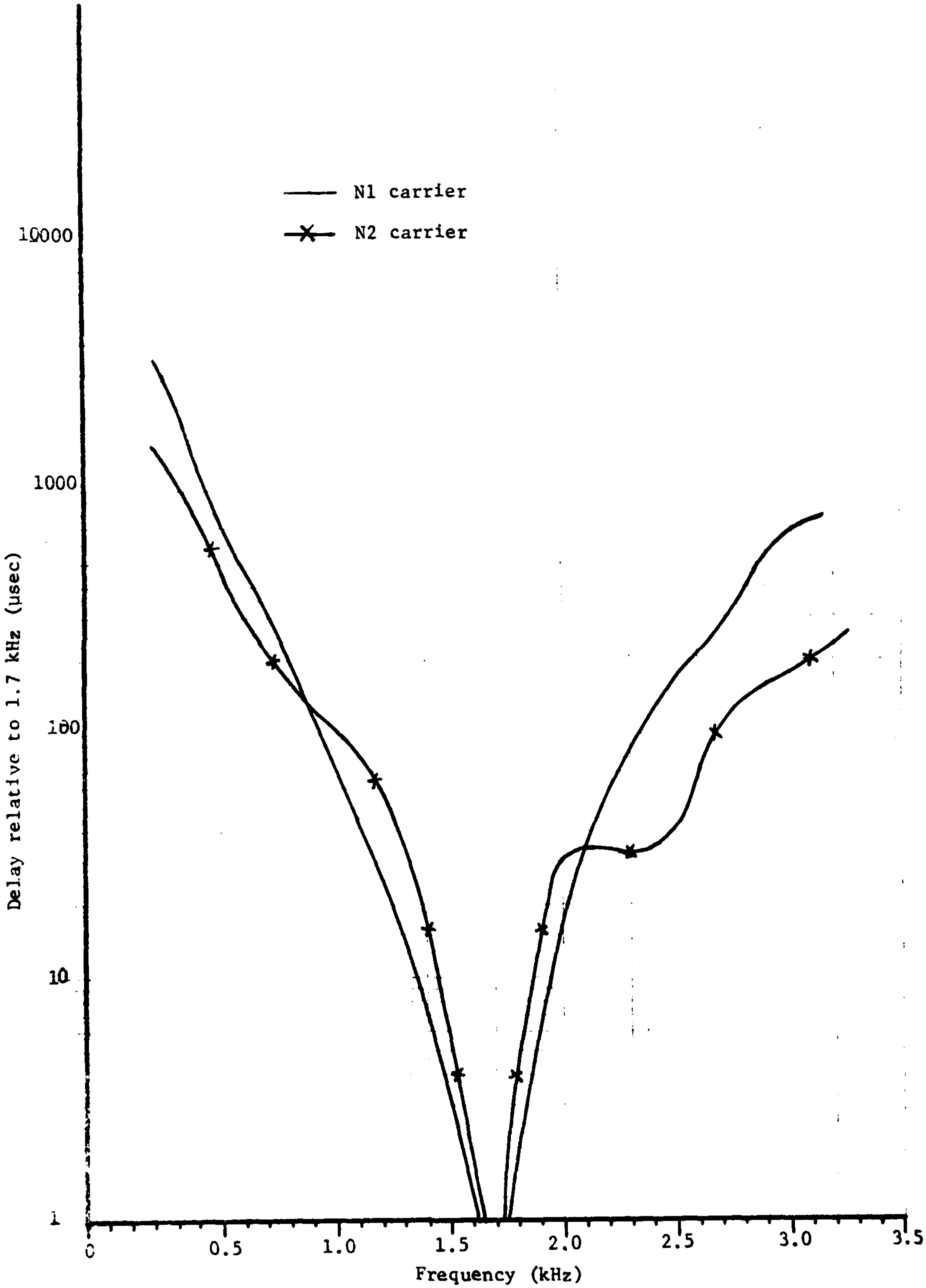


Fig. 10.--Carrier System Phase Distortion Characteristics

signal pulses under certain circumstances. As discussed previously, crosstalk is usually controlled by compandors in telephone networks. In general, these compandors have a transfer characteristic that attempts to force the digital states toward a median level and thereby reduce the signal-to-noise ratio.[58] In addition, compandors are generally designed with a time constant much too slow to follow the transitions of digital signals; consequently there is resultant distortion in width as well as amplitude. The only practical solution to this problem is to disable the compandor during data transmission. This, and each of the several other methods for automatically disabling the compandor during data transmission that have been developed to date, creates a problem when the same circuit is used for both speech and data.[59]

Similarly, echoes are problematical in data transmission for the same reasons as crosstalk.[60] The echo suppressors in general use are no better suited to data traffic than are the compandors, because they are incapable of following the transitions of the signal and they necessarily inhibit full-duplex signalling. As with compandors, the solution is to disable the echo suppressors during periods when the circuit is used for data, a procedure that has also led to the development of automatic disabling

systems activated by a data stream.[61]

Frequency translation in FDM systems as well as short-term frequency instability, can pose serious difficulties to accurate data transmission. The effect of this impairment on the demodulated data signal is virtually indistinguishable from phase distortion,[62] because varying frequency in the channel master oscillator causes shifts in the entire spectrum. Fortunately, most FDM carrier systems are of the synchronized variety in which this effect is minimized.

Long-term level variations within the carrier system are of little import to digital signals, but sudden phase and amplitude variations are quite another thing. Drop-outs of very short duration (on the order of msec) have little effect on speech because of its redundancy, but they can represent a significant portion of a data signal and can result in errors.[63] As these phenomena are unpredictable, the primary defense against their effects is the use of transmission schemes which utilize one or another form of acknowledgment of data blocks or schemes that utilize transmission redundancy.

A signal impairment that is particularly troublesome to data signals is phase jitter, the term for transmission impairments which cause individual pulses in the data stream

to be displaced from their proper position.[64] This causes high error rates in digital transmission, but the effect is little more than a curiosity to speech transmission system engineers. Jitter results from unwanted modulation of the desired signal during transmission by another signal, with resultant undesired phase or frequency changes. It can also be caused by timing errors in the pulse regeneration process[65] and is a problem of considerable magnitude in PCM transmission. Jitter is cumulative; it can be removed from data streams by retiming, but only at considerable cost in complexity.[66] Better system design to minimize intermodulation reduces jitter in analog systems, and considerable improvement in this parameter has been observed on the Direct Distance Dialing (DDD) Network in the past several years.[67] In all-digital systems, accurate timing at all points in the system, especially those where regeneration occurs, is mandatory. Accurate timing is also required for other reasons in such systems, so this requirement is not driven by reduction of phase jitter alone.[68]

Lest it seem that a catalog of all possible faults which may occur on a channel has been presented above, it is well to note that no less than 13 separate impairments are significant in the transmission of data,[69] and this is a list of only the most common and the most important. If

there is a single point to be made from that list, it is that digital signals are generally more sensitive to transmission degradation than voice signals, primarily because of the inherent nature of digital signals and their greatly reduced redundancy as compared to speech.

DIGITAL VERSUS ANALOG: A COMPARISON

It is apparent from the discussion in this chapter that neither digital nor analog techniques are "best"; each has advantages. Digital systems using present techniques are capable of much better signal quality over long circuits than are analog systems;[70] for shorter systems (up to, say, 3,000 miles) it is difficult to impute a clear advantage to digital techniques in signal-to-noise ratio.

Digital communications involve more complex equipment than analog systems, which has been a significant disadvantage until quite recently. The integrated circuit has ended the component-by-component design of circuits and systems, with the result that there is now little to choose between either method vis-à-vis complexity. The internal complexity of the integrated electronics chip, after all, is at issue, not the difficulty of understanding each portion of a circuit or the expense of connecting thousands of components to realize that circuit.

Complexity previously was the prime mover behind the cost disadvantage of digital systems,[71] but the integrated circuit has eliminated this and digital systems are now being costed competitively with, and often less expensively than, equivalent analog systems.[72]

The one area in which digital systems display an overwhelming advantage is in encryption. The automatic, on-line encryption systems in use today throughout the world are digital devices which operate on digital inputs to produce digital encrypted outputs, and vice versa.[73] There is little likelihood of a technically feasible analog, on-line encryption device being developed in the near future that will not utilize digital techniques internally (i.e., it will contain its own mini-PCM system to interface a digital encryption device to an analog line).[74] It makes little sense to preserve the form of an analog signal throughout a system if that analog signal must ultimately be converted to digital form to permit encryption, unless such a procedure is clearly superior to the alternative of an all-digital or hybrid system.

Goals of the Defense Communications System were stated in Chapter II as accommodating increased record traffic volume, providing high-quality speech service over a 12,000-mile reference circuit, and permitting increased

security of voice transmissions over the system. Given these criteria, digital communications of one form or another is obviously the choice over analog. The point at which to make the transformation is only one of many decisions that must be made if this course of action is implemented, however. A discussion of trends and problems in the digital communications field is therefore undertaken in Chapter IV.

CHAPTER IV

IMPACT OF DIGITAL TRANSMISSION

INTRODUCTION

The shift from the more familiar frequency domain into the time domain brings with it a different approach to problems and reveals some new problems which have not previously been addressed in detail because they are relatively unimportant to analog transmission. This was demonstrated for transmission impairments in Chapter III.

This chapter is intended to summarize the present status of major research and development in the field of digital transmission and to report on operational results that researchers and operating agencies have observed. The chapter emphasis is on the differences which exist with digital transmission as compared with analog transmission. The reader is assumed conversant with basic communications system theory and practices, and no attempt is made herein to review or explain these fundamental areas.

MEDIA

LOS MICROWAVE

The backbone of both military and civilian telecommunications systems remains, as it has for the past decade, line-of-sight (LOS) microwave transmission.[1] It can be made highly reliable, and its available bandwidth permits the use of high-speed digital data streams as modulating signals. LOS microwave is a dispersive medium which may be classified as conditionally stable for data transmission. The three primary time-varying effects which cause short-term variations in path characteristics are rain attenuation, multipath distortion, and deep fades.[2]

Rain attenuation, appreciable above 10 Ghz, is caused by all types of precipitation and is a function of both rate of precipitation and frequency of the microwave signal. A considerable body of statistical data exists on the frequency of rain over paths of interest, with the result that it is usually minimized by use of diversity techniques. The problem in digital transmission arises in the need to examine the precise mechanism of diversity switching and to determine whether it is capable of responding in a manner suitable for digital signals or whether it will require modification as to attack time and switching selection.

Multipath distortion manifests itself in very rapid fluctuations of signal level and instantaneous phase at the receiver. It has been demonstrated that multipath effects follow a Rayleigh distribution and thus can be predicted within reasonable bounds of certainty.[3] Frequency diversity is the method of choice for overcoming the effects of multipath for two reasons. First, there is virtually no correlation of fading between frequencies separated by 160 mHz or more. Second, this method has been developed to a high level of reliability.[4] However, the instantaneous phase variations which accompany multipath reception can be present even in the absence of fading of sufficient magnitude to activate the diversity switching equipment, resulting in time-varying phase distortion in the data stream.

Deep fades along the LOS path are the result of variations in the refractive index of the atmosphere and are also generally corrected through diversity techniques. As with precipitation, the diversity switching criteria and methods require examination to ascertain their suitability for use with digital signals.

It must be emphasized that these path parameter variations are not unique to digital signals (nor, indeed, to LOS microwave). They have been recognized for some time, and their effects have been minimized through the use of

automatic gain control (AGC) receiver circuitry and diversity techniques. As with the basic illustrations in Chapter III, however, the cure applied to solve an analog transmission problem may be inappropriate or even exacerbating for digital transmission.

The probability of a single bit error on an LOS system is directly proportional to the signal-to-noise ratio (SNR). With proper spacing of regenerative digital repeaters, microwave systems utilizing digital transmission can operate at lower SNR's than would be the case with analog signals, owing to the ability to strip white noise from the data bits during regeneration.[5] The probability of multiple bit errors, however, is a function of the instantaneous channel characteristics and can be thought of as following a Rayleigh distribution. This effect has considerable importance in the transmission of low-redundancy digital information over LOS links.[6] It should be recognized that these effects manifest themselves whether or not the actual link modulation mode is digital or analog, as the categorization of a signal as either time or frequency domain is merely a convenient means of describing it. The one is simply a mapping of the other by conformal techniques into another space.

It has been demonstrated in operating systems that a

digital bit stream can be inserted in place of analog signals on one or more of the baseband channels of a microwave radio system (usually a supergroup frequency slot of 240 kHz width), retaining analog signals on the remaining channels. Attention must be given to the maximum permissible loading of the modulator, but, assuming the engineering has been done properly, the error rate observed on such a system has been very low. To minimize adjacent-channel interference problems, the digital stream is band-limited in much the same manner as the analog signals habitually are.[7] Inasmuch as any conversion of a system from analog to digital form will necessitate a period of hybrid operation, it is worthy of note that LOS microwave has proved the feasibility of this concept and is in daily use.

TELEPHONE CABLE

Although LOS is the long-haul backbone of most telephone systems, cable carries the load on the shorter trunks and in the exchange plant. It cannot be considered insignificant since more than 80 per cent of the trunks in the Bell System are 15 miles or less in length.[8] In other systems the figure is as high as 94 per cent, and this includes toll trunking. Although the average length of a Defense Communications System (DCS) channel is approximately

174 miles,[9] because of its worldwide scope and specialized mission, cable still forms a significant portion of the plant, especially between the user and the DCS entry point, if that be a switching center. Also, much of the DCS consists of facilities leased from commercial carriers whose statistics are above.

Common twisted-pair, 22-gauge copper cable is suitable for one channel of voice (3-kHz bandwidth, or 64 kb/sec equivalent) using analog transmission. Present practice uses this same cable with 6,000-foot regenerator spacing for the transmission of pulse code modulation (PCM) at 1.5 mb/sec, and this could be increased to 6 mb/sec if the repeater spacing were cut to 3,000 feet.[10] That represents an increase in capacity of nearly 100-fold by the addition of regenerators at suitable intervals. As the cost of amplifiers is low compared to the cost of parallel cable runs, the result is a very low incremental cost for the increased capability.

Cable is an extremely stable medium for digital transmission, and this is the primary reason for the development of schemes such as that described above. The major shortcoming of such a system for digital transmission is the accumulative effect of time jitter; however, this can be minimized by the use of rather complex logic circuitry which

is now feasible as a result of large-scale integration (LSI).[11] The result is a stable, quiet, accurate digital system.

As with microwave, much effort has been directed toward achieving some degree of compatibility between digital and analog signals in the same cable sheath so as to avoid duplication of facilities. In this case, limited success has been achieved, with the major difficulty being crosstalking of digital signals into analog signals. The converse virtually never occurs (as might be expected from a comparison of the power spectra of the two types of signals at common transmission levels).[12]

The same cable that shows a 1 db/mile loss to VF can show a 31 db/mile loss to PCM, and this has led to the development of test techniques for cables carrying digital signals that are much different from the traditional techniques used on analog-bearing cables. It has been demonstrated that to achieve meaningful results for digital transmission, it is necessary to test the cable with a replica of the digital stream instead of a single-frequency tone or a simple reversal sequence. Several test sets developed to do just this are used in the commercial networks, together with performance analyzers which receive the test signal at the distant end of the cable, compare the

received signal with an ideal signal, and display the result in the form of a computed "degradation factor" of the circuit.[13]

Owing to the fast wave fronts transiting the cable in digital mode, the resistive fault-isolation and location techniques that are in common use on analog systems are not generally applicable. Rather, it is necessary to utilize test methods which very much approximate time-domain reflectometry to locate faults. It is also necessary to test the cable in both directions for the same pair, as an impedance discontinuity near one end (such as might be caused by water seepage) will affect transmission in one direction more than the other, owing to the attenuation of the standing wave it creates.[14] Commercial carriers have learned these lessons only after the expenditure of much time and funds, but the result has been another series of test sets especially designed to perform just these tests on telephone cable.

It must be noted that the transmission of high-speed data over telephone cable requires the interjection of a considerable amount of electronics in the signal path, in the form of regenerators, as compared to analog transmission. This is the price paid for increased information transfer. However, unless very specific marginal tests are possible on these regenerators from one end or the other of

the circuit, fault isolation will be prolonged and difficult and large quantities of good electronics will be needlessly removed for maintenance.[15] It should be possible to exercise each regenerator independently from the end of the circuit and to evaluate its performance against an absolute standard. This is difficult in practice, although some test sets do exist to perform this sort of test and to indicate the location of the faulty regenerator.[16]

COAXIAL CABLE

The coaxial cable transmission medium is gaining in popularity with commercial carriers as an alternative to crowded microwave bands in dense areas and because of the stable, wideband characteristics it displays. In military-owned portions of the DCS, it is most frequently utilized for submarine cable routes.

Coaxial cable is more stable and free of noise and fading than any radio system. However, compared to a similar length radio circuit, its very high transmission loss necessitates the use of repeaters at comparatively frequent intervals. The phase transfer function of coaxial cable is a smooth, slowly-varying, and predictable function of frequency which minimizes the requirement for equalization changes.[17]

The bandwidth obtainable in a coaxial cable is a function of the cable size and the spacing of repeaters, and it is usually chosen as a result of careful economic trade-off analysis. Bandwidths of 1 gHz are currently obtainable,[18] and at least one experimental PCM system operates over coaxial cable at a 224 mb/sec data rate.[19]

The same caveats expressed regarding the testing of telephone cable systems apply to coaxial cable. However, because it is a wideband medium, techniques such as time-domain reflectometry are not as uncommon to technicians who work with coaxial cable as to those more accustomed to telephone cable carrying telephone signals.

Digital and analog signals have been tested over a common coaxial cable for compatibility, and the results were excellent.[20] The method used is generally the same as that for LOS microwave; that is, the replacement of one or more frequency division multiplex (FDM) group or supergroup slots by a digital data stream. Two-way hybrid transmission is also feasible using the usual analog method of allocating certain frequency bands for each direction of transmission.

SATELLITES

The use of satellites as communications relays is increasing rapidly, especially in the DCS, where satellites

are envisioned as a primary long-haul medium in the next decade.[21] The discussion of satellites in the DCS is limited to those areas that are unclassified, but this does not severely constrain the technical comparisons of analog and digital operations.

The characteristics of the satellite path are quite similar to those of microwave radio, with the exception of selective fading, which is not a problem with satellites. Because of the extreme path length, the path loss is quite high and requires the application of much more sophisticated and sensitive electronics than is the case with microwave.[22]

Satellite systems paths, for the most part, are extratropospheric. This reduces the dependence of path characteristics on atmospheric characteristics, as much of the path lies in the near-vacuum of space. This is especially true of synchronous satellites. There are effects, however, which cause time-varying alterations to the signal path:[23]

1. Rain attenuation can be significant on satellite paths if the fall rate is very large, as in the tropics. This attenuation is slightly less for high look angles than for low but it becomes quite severe for all angles when the antenna is enclosed by a radome upon which the rainfall sheets.

2. Variations of path noise over time, and thus the SNR, occur as a result of conjunction of the satellite with celestial emitters of radio energy (stars, sun, etc.) or conjunction with another satellite.

An additional problem that may prove troublesome on links connecting computers occurs with asynchronous satellites, that being outages due to hand-over as satellites enter and leave the view of the earth station. These outages may range from a few minutes to several hours depending upon the distribution of satellites in orbit and the location of the earth station; also, they may vary from one hand-over to the next for the same reason if the satellites are randomly dispersed in orbits.[24]

Whether utilizing analog or digital techniques, the susceptibility of satellites to jamming and other electronic countermeasures (ECM) must be considered. This susceptibility is largely due to the necessity for the satellite antenna beam to subtend an angle which includes all earth stations seeking to use the satellite simultaneously. For a synchronous satellite, the earth subtends an angle of approximately 18°,[25] and a compromise must be reached between this value and the minimum angle that may be feasibly attained in consideration of the above requirement. Irrespective of the measures taken in military systems to

minimize jamming vulnerability by altering the distribution of satellites and earth stations, it must be remembered that circuits leased over satellite paths from commercial carriers may not incorporate these features, although Intelsat IVS uses a spot-beam technique[26] which focuses the antenna beam on high-traffic areas. This also helps to increase the satellite's capacity.

These problems are germane in this discussion, as there exist many systems (classified in their details) which serve to minimize the vulnerability of the satellite to ECM by means of signal-processing techniques. These systems generally trade bandwidth for speed and require complex logic.[27] For precisely that reason they are more adaptable to digital signalling than to analog. This factor should not be forgotten in making the selection of a medium.

Data signal timing can become a troublesome problem with satellite systems as a result of either the considerable path delay associated with synchronous links or the Doppler shift found with asynchronous satellites. These effects are much more of a problem to digital mode than to analog, and their effect should be examined on a case-by-case basis. Delay can be countered by retiming or by other variations of synchronization. Doppler effect, which with satellites is quite predictable, can be countered via a

variety of signal-processing techniques, to include retiming in certain circumstances.[28] The net effect of these impairments is to increase the complexity of the earth station, which is already considerable, beyond that required for analog transmission.

Another satellite development is the doubling of satellite capacity by use of cross-polarization techniques within the same frequency band.[29] This would permit utilization of the same link for two groups of signals, one vertically polarized and the other horizontally polarized. This system is not yet in use; however, it could have serious disadvantages for digital transmission, depending upon the amount of rotation experienced by the field vector of the digital carrier wave en route to and from the satellite. Such rotation of the field vector is quite common in terrestrial communications and results in crosstalk from one polarization to another.[30]

Digital-mode satellites are in use in the Defense Satellite Communications System, Phase II (DSCS II), and future military satellites promise to be exclusively digital.[31] For these reasons it can be expected that ground terminal equipment will become optimized for digital mode, and the problem of compatibility with analog transmission on the same path should not arise. A demand for such

compatibility would require considerable sophistication aloft, and the effectiveness of such an approach is dubious.

HF RADIO

The use of HF radio for backbone trunking is decreasing rapidly throughout the DCS. HF circuits are now used nearly exclusively for backup of primary paths routed via other media.[32] It is for precisely this reason that the suitability of HF for digital transmission be examined, for it is not inconceivable that in conditions of general war or catastrophe it may be the primary long-distance medium available, especially outside the Continental United States.

In addition to the problems of noise and fading which affect analog HF signals, two additional path impairments are important in digital transmission. First, HF paths suffer from dynamic delay distortion that is caused by frequency-selective refraction in the ionosphere. This effect is time-varying and unpredictable.[33] Second, the constantly changing altitude of the refracting layer of the ionosphere results in a constantly changing path length which, in turn, results in a non-constant instantaneous data arrival rate at the receiver. This manifests itself as intersymbol interference and makes the utilization of synchronous transmission exceedingly difficult.[34]

AD-A089 534 ARMY COMMAND AND GENERAL STAFF COLL FORT LEAVENWORTH KS F/G 17/2
CONVERSION OF THE DEFENSE COMMUNICATIONS SYSTEM FROM ANALOG TO --ETC(U)
JUN 74 R A STANLEY

UNCLASSIFIED SBIE-AD-E750 001 NL

2 OF 3
AD A
089534

Methods exist to minimize these degrading effects on the data signal, however, and HF links have been reliably used at rates of 4800 bps with an error rate of 10^{-2}. (Considerably lower error rates are experienced as the data rate is decreased to 2400 bps.)[35] Although not good by computer standards, the magnitude of error 10^{-2} is suitable for digitized voice and for some types of teletype transmission.[36]

Although not in favor in the DCS at the moment, the use of HF should not be summarily excluded for digital signals. It is simple, economical, and easily transported. While United States Forces have opted for more sophisticated systems, the armed forces of several nations, notably the United Kingdom, operate worldwide communications systems with an HF backbone at respectable reliabilities.[37]

TROPOSPHERIC SCATTER

The use of tropospheric scatter radio in the DCS has increased markedly over the past decade, largely due to its ability to span relatively long distances without intermediate relay sites. This proved invaluable in a combat environment in Vietnam and is well suited to routes over water, such as the link from the United Kingdom to Newfoundland via Iceland, which is comprised of mixed tropo and LOS links.

Tropo is not widely used in commercial systems, largely due to the length of the average commercial trunk, which is quite short, as has been shown.

Tropospheric scatter propagation is affected by all the impairments which affect LOS microwave, but the primary effect of interest in data transmission is frequency-selective fading, which appears in the time domain as intersymbol interference. The fading mechanism is the predominant impairment at high data rates and negates the usual assumption of flat Rayleigh fading over the signal bandwidth. From the system standpoint, this has the effect of imposing a floor on the minimum achievable error rate, which is unalterable by varying the SNR. This is an extremely serious impairment for the transmission of data.[38]

At lower data rates the Rayleigh fading assumption holds and, coupled with random noise injection, becomes the primary path error mechanism.

Owing to the wide dispersion of tropo systems in the DCS, much study continues to minimize the effect of these impairments on the transmission of digital data streams. At the present state of the art, tropo is only about 50 per cent as efficient in the digital mode as in the analog mode,[39] and it is the transmission medium most seriously affecting the introduction of digital signals on a large-

scale basis in the DCS.

The methods presently available to minimize path effects in tropo are:[40]

1. Use of all available statistical data for the design of an optimized system. Although feasible, this requires that different receiver transfer functions be developed for each path, which further requires a substantial investment in engineering and manufacturing of many low-volume items.

2. Use of adaptive receiver techniques which alter the instantaneous receiver transfer function by determining the instantaneous channel characteristics. This approach requires extreme sophistication in equipment and remains to be demonstrated as being feasible on a large scale.

Present techniques in tropo provide acceptable performance with data streams up to 3 mb/sec, but the use of higher speeds in the digital mode is constrained by the excessive radio frequency (RF) bandwidth required and by the path limitations already discussed.[41] For comparison purposes, this data rate is equivalent to 24 voice channels using standard PCM techniques.

MILLIMETRE WAVES

Although known for some time, the use of millimetre

wavelengths for information transmission has only recently become technologically and economically feasible, primarily through the development of Impatt diodes and manufacturing techniques which have made waveguide affordable. Millimetre-wave systems offer extremely wide bandwidths (because of the extremely high carrier frequencies used), relative freedom from RF interference (as this portion of the spectrum is not heavily used at present), and reasonable cost.[42]

Millimetre-wave transmission through the atmosphere is subject to heavy attenuation from precipitation, water vapor, and oxygen absorption. Because of the very high path loss experienced in air, only short distances between relay sites are possible, usually on the order of 10 miles or less. The nature of the path impairments is such that space diversity is the preferred method of minimizing their effects, with the result that two systems must be built, effectively, in parallel.[43]

However, the transmission of guided millimetre waves in the frequency range 40 to 110 gHz offers considerable promise. It is now possible to construct practical circular waveguide for millimetre-wave transmission in the TE_{01} circular mode.[44] As compared to the atmosphere, waveguide is a very low-loss medium, with losses resulting primarily from resistive loss in the guide walls and from standing

waves and mode-switching. These latter effects are also sources of distortion.

Using the TE_{01} mode in circular guide, any deviation from perfect cylindrical geometry will result in mode-switching. This is compensated by the use of mode filters and signal regeneration, with a resultant guide loss of 2 to 3 db/mile (without amplification). Repeaters, therefore, may be located at 20-mile intervals and operate at low power levels (0.1 watt). Regeneration being a prerequisite for practical transmission, digital modulation is virtually obligatory for millimetre-wave transmission in waveguide.[45]

Largely because of the large bandwidth available, millimetre-wave systems are being extensively developed.[46] The economics of this medium, however, make its use cost-effective only at very high utilization factors. As yet, even the busiest route in the Bell System has insufficient traffic volume to justify the installation of millimetre-wave equipments (or their bandwidth equivalent).[47] Military planners should bear this in mind, although the stability and security of this medium may well be factors which outweigh cost-effectiveness.

In anticipation of increasing traffic demands, millimetre-wave systems are presently being tested by the Bell System in the United States and the Deutsche Bundespost

in Germany. The latter system uses no repeaters for a 26-mile waveguide route and carries 500,000 voice channel equivalents.[48] The Bell system has a capacity of 240,000 voice channel equivalents and is under study for increasing this number through signal processing techniques. Bell officials state that an operating millimetre-wave system can be fielded by 1978, and they expect this to become the high-density trunking medium of the 1980's[49] as increasing traffic makes it more economically attractive.

OPTICAL TRANSMISSION

Available bandwidth being a direct function of carrier frequency, optical transmission offers the promise of a stable medium for extremely wide bandwidth service (such as giga-bit inter-computer links).[50] The narrow beams obtainable with lasers and the present lack of a requirement to license the transmitter offer a high degree of privacy as to both the content of the transmission and the very existence of a communications system.

Although widely advertised as the "coming thing" for digital communications, optical systems are more expensive than millimetre-wave systems of the same bandwidth and thus will not prove economical until much later than millimetre-waves. The best current opinion is that optical systems

will not be used in operating systems prior to 1990.[51] Thus they are not discussed further herein.

COMMON CARRIERS

It is frequently desirable or necessary in the DCS to provide communications routes and services by leasing from a common carrier. This is especially true in the Continental United States, where the military is prohibited by law from providing services that can be provided to the requisite standards by common carriers. A similar situation frequently obtains overseas as a result of agreements with host governments concerning use of their communications facilities.

Although common carriers use the already-described electrical means for the transmission of information, it is usual in the DCS to consider them as a separate medium of communications, and that distinction is preserved here. This treatment of the common carriers is convenient, as when one leases circuits and services from them it is necessary to specify many of the same items that would be specified for a purchased system.

Foreign common carriers nearly always take the form of a government owned or operated post and telegraph monopoly. Outside the United States they generally adhere to

technical standards established by the CCITT, but rate structures and tariffs are a national matter and vary widely from nation to nation. For this reason foreign common carriers are not discussed in detail herein. In the main and within the framework of the exceptions above, everything discussed in regard to domestic carriers applies to the foreign carriers. As in the United States, the PTT's of most developed nations are expanding the data transmission networks within their countries.[52]

The United States, virtually alone among the nations of the world, permits competition in the communications carrier field, albeit under close government regulation. Until very recently the only true nationwide communications systems have been those of the Bell System and General Telephone, the former being far and away the larger. The Carterfone decision[53] and the rapidly growing demand for data transmission services, however, have spawned several competitors in the data transmission field, and the announced policy of the government, as a user of these services, is to encourage competition among the carriers.

The Bell System, by judicious insertion of digital streams into the FDM channel or group slots, has provided data service over its existing facilities for many years. Thus the common denominator of data has been the

3-kHz channel. For service requiring bandwidths beyond the capacity of a voice channel, it was necessary to lease a group slot or a supergroup. This system is still widely used in the Bell System and most other telephone companies. Typically, the carrier, in accordance with the user's specifications, provides the leased circuit with all necessary conditioning equipment.

The Bell System has found, however, that its ability to integrate digital signals into its predominantly analog system has become limited. As a result it has announced its Digital Data System (DDS), which is "functionally discrete from but physically integrated into the existing Bell System network."[54] This system is designed to accept signals at 2.4, 4.8, 9.6, or 56 kb/sec between two points, with synchronization provided by the network clock. The system neither currently provides for switched service nor is capable of handling digitized voice. It is designed to provide > 99.5 per cent error-free seconds or better at all data rates.[55]

Customers who require switched data service which can be handled over a voice channel (up to 9.6 kb/sec, with currently available modems) can utilize the Direct Distance Dialing system to pass their traffic. The performance of this network in the handling of digital traffic has steadily

[illegible] over the past years,[56] and it is the primary means for the exchange of switched data in the United States. Tariff difficulties abound with this system, however. The minimum billing unit for the DDD system is three minutes connection time, which is significantly more than the average data terminal connect time. Thus, use of the DDD system for this type of information transfer results in over-expenditure for communications than would be true if only actual time used were billed.[57] In addition, Bell has added a charge of from 100 to 650 per cent over previously existing charges on the DDD net for all calls which access an "automated information system," which has been interpreted to mean any sort of computer.[58] Thus, the cost of using the DDD system for data transfer has become much more than was envisioned by those users now operating over it.

In the area of digitized voice, Bell has used PCM carrier in local exchange areas for more than 10 years.[59] Its T-1 carrier system is a 24-channel PCM system using an 8-kHz sampling rate and an 8-digit PCM word, including signalling and synchronization. The data rate is 1.544 mb/sec. Although developed primarily for exchange plant use over telephone cable with digitized voice, it is capable of a digital error rate on the order of 10^{-8} when used for the transmission of externally generated data

streams.[60] The T-1 system now accounts for 25 per cent of Bell's total exchange plant circuit miles, and this share is steadily growing.[61]

The success of the T-1 system has led to development and fielding of the T-2, a trunk carrier system of 96 channels capacity. The T-2 can be thought of as equivalent to four T-1's, but its system length is up to 500 miles, with error rate less than 10^{-7} on at least 95 per cent of the 500-mile routes.[62] This system is presently in operational use and is expected to displace older analog systems as they are retired.

These digital carrier systems exhibit virtually perfect load factors and excellent channel characteristics.[63] They are compatible with all existing Bell carrier systems at channel level and with all digital systems at baseband. The Bell system has arranged its digital lines in a hierarchy such that a T-1 output (1.544 mb/sec) becomes one input of four to produce a 6.312 mb/sec rate at T-2 level. The T-2 level, in turn, becomes an input to produce a T-3 level, and so on, as was the case with channels, groups, supergroups, mastergroups, and jumbogroups in the analog system.[64]

More important to the user, however, is the ability to access this system at any level with a data stream. At

present it is impossible to transmit data reliably over analog telephone channels at rates much above 9.6 kb/sec. However, the channel rate of T-1 carrier is 64 kb/sec, which can accept a compatible data stream directly when the voice digitizer is bypassed. This represents an efficiency improvement of 667 per cent and saves the rental fees on seven analog circuits which would have been required in that mode. Further, digital channels can be stacked to provide service rates at multiples of 64 kb/sec, quite unlike the case with analog channels, from which the next step is a group frequency slot.[65]

The other major telephone company of the United States is the General Telephone and Electronics Company (GTE), which operates many exchange areas throughout the country. Most of its long-distance trunking is provided by the more extensive Bell System, with which it interfaces quite effectively. Bell and General do not compete within an area, so the user has no choice of systems. Because of their extensive interface, however, virtually everything said above about the Bell System, with the exception of the DDS, applies equally to the General Telephone network.

Owing to this interface, the transition from the General System to the Bell System is virtually invisible to the user, and the equipments used are compatible. The PCM

carrier system used in the GTE network which corresponds to the Bell T-1 is the Lenkurt 91A, which is designed for use over cable, as is the T-1.[66] It can reasonably be expected that this system will develop using the same hierarchies and standards as the Bell System, and thus the Defense Communications Agency (DCA) can treat it as part of that system (although not for contracting purposes, obviously).

Both Bell and GTE are telephone system operators who have become providers of data service rather by default. The growing gap between the demands of data users for transmission capacity and capability of these networks to provide was a natural scenario for the growth of competing carriers, and several organizations were created to provide this service. The major competitors to the telephone systems at present are the Western Union Telegraph Company (WU), MCI, and DATRAN.

Western Union, whose bread and butter is the provision of teletype service, has created a concept which closely resembles the Bell hierarchy for the DDS, with the exception of an added time-division multiplex (TDM) stage to provide user services at speeds up to 1200 bps, and then multiplex these speeds onto a 2.4 or 4.8 kb/sec DDS-style line. Western Union presently provides 20 per cent of all local data traffic distribution nationwide, and 36 per cent

in 20 major cities.[67] They are presently expanding this service, which uses specialized techniques to minimize the impairments found on telephone exchange systems, to attract still more customer interest. At the same time, WU is upgrading and modernizing its multipoint and switching services.[68]

Although MCI and DATRAN provide local distribution, they concentrated on the high-volume, long-distance market initially, with switched data service to follow. These systems are specifically designed for the transmission of digital data and follow generally the Bell digital hierarchy. DATRAN, for example, uses LOS microwave exclusively for long-haul trunking, with the links engineered for an exceptionally high fade margin of 50 db. The systems are transparent to the user, who must furnish whatever terminal-provided services he desires (e.g., ARQ). Neither of these systems provides truly uniform coverage of the nation, their primary routes being concentrated along the routes of highest data traffic.[69]

The primary attraction of the data carriers has been a tariff below that offered by Bell, which has accused the newer companies of "skimming the cream" since Bell is required by its tariff to serve all locations in the nation (together with the interfacing smaller companies, such as

GTE, to form a national communications network). A recent rate reduction by Bell for the services most popular on the MCI and DATRAN routes[70] may presage the commencement of a price competition in the provision of these services. Provided the services themselves are neither degraded nor discontinued, this can only help the consumer of the service if it is allowed that no company will offer a tariff it cannot continue in the long term.

MULTIPLEXING

Chapter III has indicated the effect on signal characteristics and impairments of the various types of multiplexing and modulation that are available. However, to make an intelligent choice of a mode of operation of a digital system, it is necessary to consider the means of comparison to be used, the philosophy of multiplexing and routing, and the need for synchronization.

COMPARISON OF MODULATION SCHEMES

Unhappily, there is no "best" modulation method for digital systems. In addition, the comparison of competing alternatives is fully as difficult as the comparison of costs.

One model has been developed[71] which rates the modulation systems as a function of relative power required

to produce a given SNR. The results obtained from this model when one uses a reasonable set of assumptions for conditions likely to obtain in the DCS are shown in Figure 11. Given a transmission system which maintains a given SNR, selection of the system which requires the lowest relative power to attain that SNR after modulation/demodulation implies selection of the lowest-cost system if one assumes that the incremental cost of a given power increment is constant for all modulating systems. Given wide use of LSI techniques, this may be an excellent assumption in the near future, although it is not in the case of discrete device technology. Conversely, given a modulation system that requires a certain power to attain the desired SNR, one can select the least costly transmission system to provide the desired characteristic.

An approach used by Bennett and Davey[72] compares the various systems on a "figure of merit" basis by imputing arbitrary values to the systems' noise vulnerability, complexity, cost, etc. This system is inexact but considers most of the significant variables. It may prove especially useful when used in conjunction with a model such as that described above.

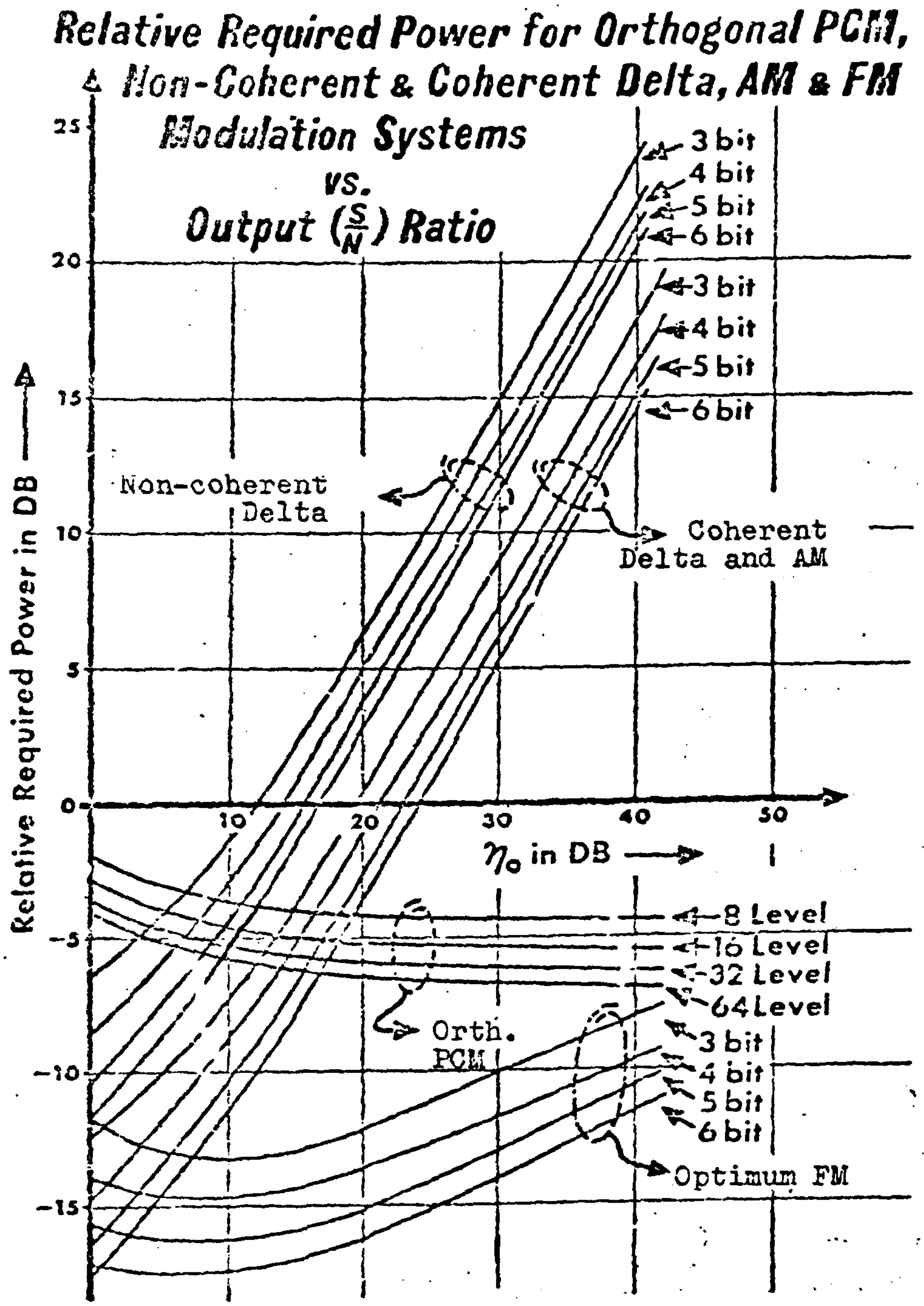


Fig. 11.--Comparison of Modulation Systems

MULTIPLEX PHILOSOPHY

Multiplex technology originally developed as a method to save on long-haul cable and radio installation along parallel routes in telephone networks by increasing the capacity of a single such path. The philosophy of its use changed little until recently, when several investigators suggested that this approach may not be optimum for data networks.

The largest and most inefficiently utilized investment in physical cable plant is that in a telephone system exchange plant.[73] Considerable effort has been expended to maximize the utilization of toll trunk facilities, but little of this has been applied to the exchange plant. As anyone who has one in his home knows, utilization of the average telephone is quite low and, by extension, so is utilization of the dedicated cable plant that connects it to the central office.

The use of the DDD network to handle switched data serves only to exacerbate this problem, for the nature of these data is primarily bursts of information at relatively high speed, separated by long intervals of waiting time during which no information transfer occurs.[74] If several data terminals in the same geographical area wish to access the same computer simultaneously (a not-uncommon situation),

each must be connected via its own dedicated telephone line, will be subjected to the 3-minute billing minimum, and will be forced to accept incredibly low efficiency on the line while connected.

The solution to these problems is often to place multiplex terminals close to the user, so as to more efficiently utilize the exchange plant. By the installation of multiplexers and/or concentrators at the user end of the circuit, utilization can be improved significantly without incurring additional cable costs. This is significant, especially as the rising cost of copper and the falling price of LSI functions have combined to make digital multiplexers less expensive than cable runs for many lengths of circuits.[75]

A further refinement of this approach is the use of random-access techniques to create a situation whereby interaction is permitted among the users of a channel in such a manner that a non-active user consumes no channel capacity. In this instance the number of interactive users that can be supported by a single channel under typical loading conditions of a data terminal is approximately 160.[76]

Although this study is limited to the DCS backbone, it is significant to mention here that a major decision in

multiplex philosophy concerns the location at which digitization of analog signals takes place within the system. Traditionally, this has been done in the multiplex equipment at a nodal point, after the signals have passed through a central office and a large quantity of exchange plant cable. It is quite likely that this is not the optimum scheme for use in an all-digital system, especially viewed in the light of rapidly decreasing costs for LSI devices, such that a PCM voice encoder can now be fabricated on a single chip.[77]

One of the primary reasons multiplex technology and philosophy developed as they did is that until very recently the complexity of the multiplex equipment made it much more costly than the exchange plant required to connect it to a user. This is no longer the case for circuits of more than a few thousand feet.[78] As the voice digitizer is moved closer to the user end of the circuit, the cost of subscriber equipment rises and the cost of the exchange plant falls as its utilization rises. The exact location of the break-even point is extremely technology-dependent, but it is important to locate it because the entire philosophy and hierarchy of the network depend upon it. At least one study has concluded that by 1980 digitization of voice at the user instrument will be feasible.[79] That study determined further that although such a step will more than double the

cost of a telephone instrument, the overall system cost will be reduced dramatically owing to savings in multiplex and outside plant.

SYNCHRONIZATION

Accurate timing is mandatory in any time-domain system, for it is the position of the pulses in time that conveys the desired information. As in the other areas of multiplexing, there are several approaches to this problem.

Hierarchical clocking has been used in most TDM systems in the past and is used in several newly developed systems. In this scheme the system is synchronized by a stable clock which locks less-stable clocks located at hierarchically lower positions in the network to the master time reference. The disadvantages of such a system are numerous:[80]

1. The system must be extremely reliable, as a fault can easily destroy the timing of the entire network.

2. To equalize timing delays, there is a requirement for a large amount of storage at each multiplex point.

3. The start-up costs are high.

4. Transmission delays must be precisely matched.

Another system, known as phase averaging, averages the frequencies at each location. Although less demanding

of reliability than hierarchical clock systems, it still requires precise control of phase and delay.[81]

An alternative to both the above is the use of stable clocks at each location in the system. This approach is feasible, and storage can be used to compensate for delays by correcting the bit stream generated by the clock. Only 10^3 bits are required daily for correction of a clock with short-term (24-hour) stability of one part in 10^{-10}. Such stability is readily attainable with atomic clocks; the stability of a cesium beam standard is 1 in 10^{-12}. Atomic clocks, however, are extremely expensive, and economy demands that they be shared between locations. This reintroduces the problem of reliability. Additionally, to maintain synchronization, bits must occasionally be deleted or added. Unless the data stream is frequently reframed, this will cause chaos with data streams.[82]

Bit stuffing is a technique Bell System developed in an attempt to overcome the deficiencies of these synchronization systems.[83] Being asynchronous, it requires no synchronized coder clocks, and it does not lose bits. Basically, it utilizes a line data rate somewhat lower than that used at the multiplexing points. A coder examines the incoming data stream, and additional bits are "stuffed" into the signal to raise the data rate to that required by the

multiplexer/demultiplexer (muldem). This is done according to a prearranged scheme so that the added bits are then removed at the output. Because of its independence of stable clocks, bit stuffing has been adopted as the synchronization system in the T-1/T-2 hierarchy.[84]

A final alternative is the use of differential coherency, whereby synchronization is derived from the received signals without use of a separately transmitted or generated time reference signal.[85]

Research in the field of synchronization continues in an effort to discover a simpler and less expensive system than those described above. At the moment, bit stuffing appears the most promising approach for communications systems which must operate under adverse conditions.

QUALITY CONTROL

Criteria and standards associated with existing systems tend to be perpetuated as the means of measuring the efficacy of newer systems primarily because of the familiarity of the standards and their use and interpretation. It is well to reflect, however, that these standards were not decreed from on high as absolutes but, rather, were developed over a period of time in an attempt to measure the degree to which the system performed its design function.

These standards reflect both the function of the system and the means by which it accomplishes that function and, as a result, may not be well suited for use with other, different systems used for different functions or using different means to the same end. Establishment of digital networks requires reevaluation of existing system standards vis-à-vis their appropriateness to the signals being processed.

The primary areas of quality control in communications systems are the establishment of standards, the measurement of system criteria for comparison with the standards, and the procedures for instituting corrective action in the event comparison is unsatisfactory. Independent efforts have demonstrated that the standards and measurement techniques presently used to evaluate analog systems are generally inappropriate in the digital case. The implications of this research are discussed in this section.

STANDARDS

The Bell System practices and CCITT standards developed to determine the adequacy of the telephone for the communication of analog signals were later modified to include criteria for the transmission of digital information in analog form. These standards are unsuitable for an all-digital network because they measure phenomena in the

frequency domain. They are not entirely appropriate to the task even when mapped into the time domain.

System standards must be appropriate to the requirements of the terminal equipment connected to the system. They must also provide an overall measure of quality for the majority of users; in other words, not be established at the level of the most demanding user on an arbitrary basis. Users who require service above the normal system standards should be required to provide the means to attain that service, either through special engineering of their circuit or through their terminal equipment.[86]

Research has shown that bit error rate is a suitable performance measurement for most purposes. The error rate requirements of the various types of military communications, however, are quite different:[87]

1. Digitized voice intelligibility is practically independent of bit error rate until the rate approaches 10^{-2}, at which point intelligibility decreases rapidly, with speech becoming poor at an error rate of 10^{-1}.

2. The accuracy of teletype transmission is limited by the physical accuracy of the teletypewriter machines used to terminate the circuit, which is an error rate of approximately 2×10^{-5}.

3. Similarly, the error required for communications

between computers, or between terminals and computers, is limited by the accuracy of the least accurate machine on the link. The internal error rate of computers is in the range of 5×10^{-12},[88] which is considerably lower than can be provided by existing communications systems.

It is seen, then, that only in the case of computer-to-computer communications is the communications channel the limiting factor in attainable error rate. Inasmuch as reducing the error rate is an expensive process, analysis is proceeding on the use of forward and rearward error-correction schemes that will permit the attainment of a lower effective error rate than provided by the bare channel.[89] Coding also is expensive, and much of this work is directed toward locating the break-even point between coding and channel refinement.

PERFORMANCE MEASUREMENT

The accurate measurement of an analog channel requires the determination of at least 13 different parameters and a detailed analysis of their synergistic effect upon the communications quality of the channel.[90] This is clearly not the best solution, but it has evolved as more and more requirements were levied upon the telephone system.

It will be recalled from the discussion of signal

characteristics in Chapter III that the channel characteristics which most severely affect the digital signal are SNR and phase distortion. From the above discussion, it is seen that bit error rate is also a parameter of primary concern. Research has shown that these parameters, in fact, can be used to describe the performance of a digital channel quite accurately. It has also been shown that SNR and instantaneous error probability are related, as shown in Figure 12.[91]

By using the pulse-to-average ratio (PAR) measurement technique, the determination of channel characteristics is possible much more rapidly than with separate measurements. It is possible to automate the PAR measurements and, therefore, increase the speed of circuit evaluation. PAR results have been shown to have a correlation of 0.96 with envelope delay and circuit loss performance,[92] which makes this a valuable and accurate analysis tool.

The other characteristic of primary importance is a bit error rate. It has been shown, however, that the common method of specifying error rate is not entirely suitable, as the probability of an error is not independent of the occurrence of a preceding error unless the intervening time is very long.[93] It has also been demonstrated that the distribution of errors on operating circuits is non-uniform in time, with errors tending to occur in bursts.[94] Average

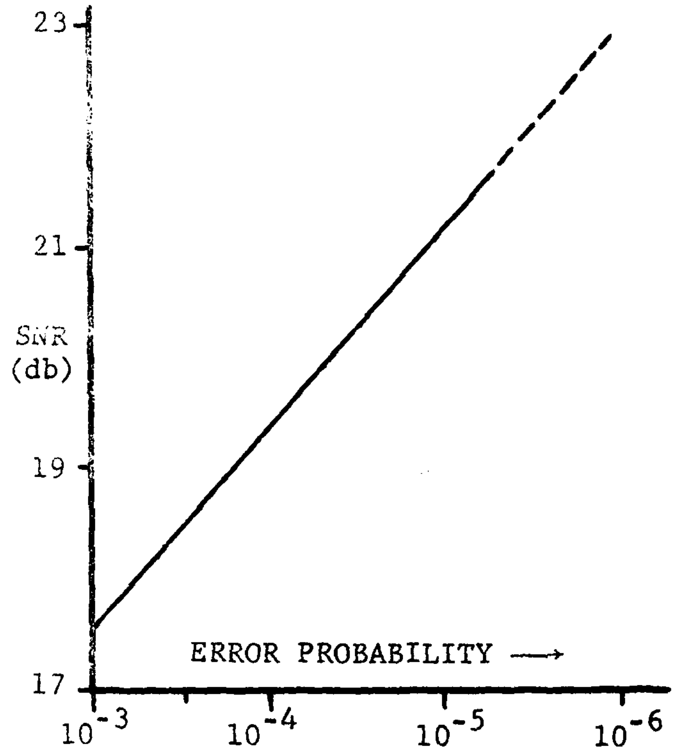


Fig. 12.--Correlation of signal-to-noise ratio and error probability for a typical medium-speed voiceband data set.

figures, therefore, are misleading, for one study results indicate that 65 per cent of their 200,000 bit messages were transmitted entirely error-free, but the remaining 35 per cent were corrupted by many errors per message.[95]

It is possible to determine the performance of a digital channel under load by counting the number of errors that occur. This procedure is used for quality monitoring in the T-1 system and can be arranged to provide dedicated monitoring of a single channel, shared monitoring of several channels, or a slow-scan monitoring of an entire channel bank.[96] The decision depends primarily upon the urgency of knowing the characteristics of a given channel. DATRAN uses a similar system, comparing errors received on each path, to determine when to activate its link-diversity switch.[97]

At present, the prediction of error probability as a function of time is inexact, and considerable study continues. A more precise model of this phenomenon will enable the design of significantly lower-error communications systems through the use of adaptive techniques. Several proposed intuitive models of the error model lack validation.[98]

FAULT DETECTION, ISOLATION, AND REPAIR

Current technical control and fault detection

procedures rest primarily on the receipt of a customer complaint as the genesis of corrective action.[99] This is inadequate in the present analog system, but no true alternative exists. Owing to the nature of the signals transiting the system, for instance, it is not possible to accurately monitor in-service channels for quality on voice links. Continuation of the present system of fault detection into a digital transmission system can lead only to net service degradation.

Fortunately, alternatives are available--and even obligatory--in the digital mode of operation. It is possible, as discussed above, to monitor the data stream on channels in service and thereby provide an alarm or an automatic reaction to a circuit failure or degradation that is beyond predefined limits. Similarly, automated PAR measurements may be taken on circuits during short out-of-service periods (on the order of seconds in duration) to identify trends circuit quality. These techniques, together with a variety of signal routing capabilities, offer the system designer the ability to utilize automatic, adaptive technical control to maintain the integrity and quality of the net under all but the most catastrophic circumstances. The same group of techniques can automatically locate the fault and thus save considerable time in that step.

Adaptive technical control is expensive, so the benefits to be obtained therefrom must be balanced against the cost.

Manual testing techniques are in much more general use, particularly among the common carriers. Bell discovered soon after the installation of the first T-1 systems that conventional fault isolation techniques were unsuited to digital lines and were frequently responsible for the removal of good electronic equipment owing to the inadequacy of the test used to evaluate it.[100] It was also found that the use of standard repetitive patterns for digital circuit testing (such as the "fox" test used in teletype) often provided spurious error indications, and the only suitable procedure for testing has been to develop a test set which generates a pseudo-random bit stream that effectively simulates the PCM bit stream.[101] Both Bell and GTE have production test sets which perform these functions, but, in spite of careful human engineering, these sets still require considerable skill on the part of operating personnel in order to properly interpret their outputs in terms of system faults.

SWITCHING

Irrespective of the manner by which they are controlled internally, all telephone system switches currently

in operational use are designed to accept only analog signals at their input/output ports. These switches utilize space-diversity switching almost exclusively, primarily using electromechanical switches to interconnect circuits.[102] The impact of this fact is that digital signal streams, prior to entering such a switch (that is to say, *any* switch), must first be converted to analog form and then reconverted to digital form at the output. Although theoretically an infinite number of such conversions is possible, the number is limited in practice by quantizing noise accumulation and jitter, thereby placing a finite limit on the number of nodal switching points which may be utilized in a digital system. Owing to the considerable capital investment represented by these switching systems, this condition will persist for a number of years.[103]

The Bell System has designed and built a prototype all-electronic, digital-mode switching system, the Number 4 Electronic Switching System (ESS), which is presently being installed for operational testing in an operating company. This switch, if proved successful, will become the primary toll switching system within the Bell System and will gradually replace existing electromechanical switches and the digitally controlled analog mode Number 1 ESS, now the standard Bell System switch.[104] This progress should be of

significant interest to the DCS, for the present Automatic Voice Network (AUTOVON) switch is a 4-wire version of the Number 1 ESS,[105] and a conversion to all-digital mode could very possibly proceed from the development of the Number 4 ESS in a 4-wire configuration.

ROUTING

The philosophy of circuit and message routing, as discussed earlier in this chapter, was developed from the requirements of the users and the technology at hand to meet those requirements. The advent of digital communications systems offers an opportunity to alter these methods in view of this new technology and user requirements, and several researchers have considered this possibility. Two of the more promising approaches are presented below.

PIERCE LOOPS

J. R. Pierce recently proposed a ring network for the transmission of data blocks.[106] The general form of such a network is shown in Figure 13, which contains A, B, and C types of modules. These are standard modules, and they perform the following functions:

A is a clock/buffer module that is used to time and close the loop. One is required per loop.

B is the entry/exit module. It is used to enter or

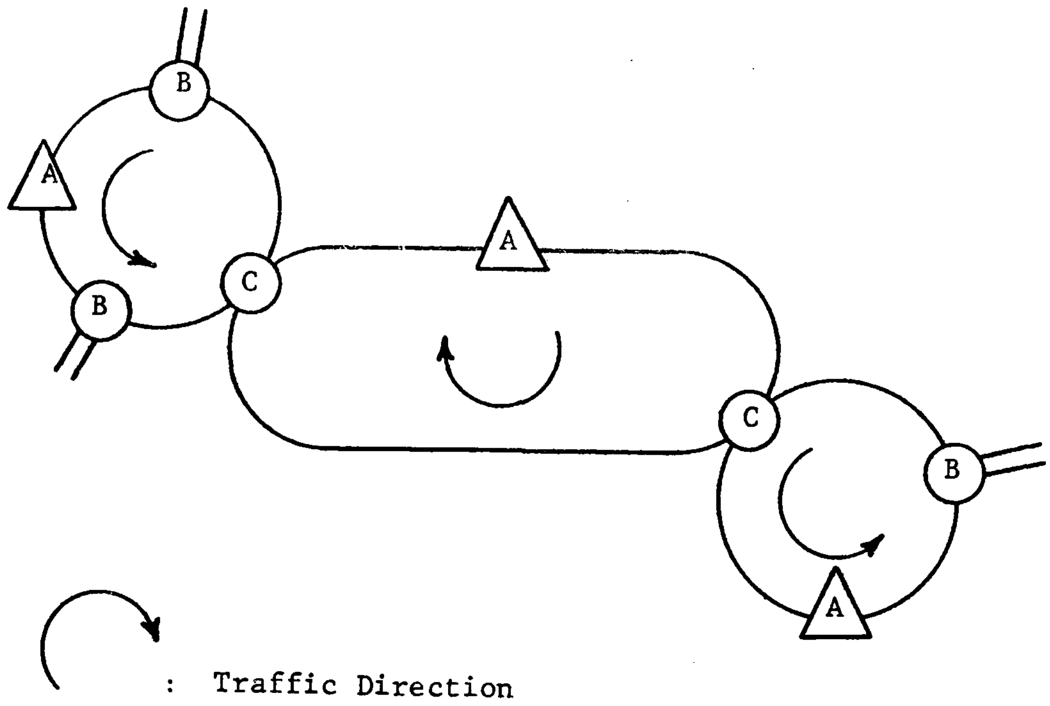


Fig. 13.--Ring Network for Data Block Switching

remove data blocks from the loop and is the means by which users are connected to the system.

C is the block used to transfer blocks from one loop of the network to another loop.

The operation of this network is quite simple. A direction of traffic flow is defined, and each data block entered onto a loop flows in that direction around the loop until it reaches the destination for which it is intended. As each data block is addressed internally, each module can examine the block as it arrives. A block not intended for a module is passed on to the next module. If it is intended for that module, it is removed from the loop and transferred to the output side of the module, which is either a user or another loop. Hierarchies can be designated among these loops to permit routing by means of derivative addressing, as is presently done in the Automatic Digital Network (AUTODIN) and in military telegraph networks. In the event of a busy condition at the destination module, the block is passed around the ring again until it can be accepted.

The ring network, or Pierce Loop, as it has been dubbed, has a number of advantages for data block transmission. Switching and routing are simple and make use of standard modules. The system adapts to line speed, and neither synchronization nor special modulation formats are

required between interconnected loops. The block formatted data are quite easy to synchronize within the loop, and the system provides rapid delivery of data blocks.

The use of this network in experimental situations has shown the concept to provide for "simple but convenient"[107] interconnection of computers or telephone carrier equipments (T-1 systems).[108] As programming is not difficult, the system is gaining favor for further study.

The primary disadvantage of the Pierce Loop is that under heavy-traffic conditions stations tend to band into classes, with the high-volume users tending to "hog" the network to the exclusion of the low-volume classes of users.[109] This situation can be overcome by appropriate network controls at a sacrifice in programming simplicity.

PACKET SWITCHING

A somewhat similar system of switching data blocks, or packets, has been developed. Known as packet switching, it does not depend upon a ring network but is adaptable to virtually any form of network configuration. A nation-wide computer network, sponsored by the Advanced Research Projects Agency (ARPA) of the Department of Defense, uses packet switching and operates as described in the extract that follows.

> The basic design of a packet-switching network, as exemplified by the ARPANET, consists of a collection of geographically dispersed minicomputers called *interface message processors* (IMPs) interconnected by many 50-kilobit/second (kb/s) leased lines. An IMP accepts traffic from a computer attached to it called *a host*, formats it into packets, and routes it toward its destination over one of the 50-kb lines tied to that IMP. Each IMP in the network receiving a packet examines the header and, making a new routing decision, passes it on towards its destination, possibly through several intermediate IMPs. Thus, a packet proceeds from IMP to IMP in making its way to its destination. The destination IMP collects the packets, reformats them into messages in the proper sequence, and submits them to the destination host computer. Throughout the process, each IMP checks the correctness of each packet by means of both hardware- and software-based error-control techniques. If the packet is received incorrectly due to a transmission error on the line, the IMP does not acknowledge receipt and the preceding IMP must retransmit the packet, perhaps over a different path. Because the network uses high-speed transmission lines and short packets, and all data is stored in high-speed primary memory in the IMPs (as opposed to disk drives and other secondary storage devices) average end-to-end transit delay for a packet is 0.1 second.[110]

Packet switching is attractive largely because of the advantage it offers the users by resource-sharing. For the current subscribers to the ARPANET, a cost reduction of three-to-one has been achieved by means of packet switching as compared to acquisition of computing equipment at each user location to perform the same functions now performed via ARPANET.

The use of packet switching permits high reliability through the use of adaptive network techniques and automatic alternate routing, and also enables the network to allocate communication bandwidth dynamically as required, rather than utilizing preassigned bandwidths designed for either the

busy hour or average load.[111]

One of the primary attractions of packet switching and Pierce Loops for data transmission is their economic advantage over alternative solutions. Although quite sensitive to computing costs, packet switching permits the construction of a network in which there is virtually no correlation of cost and distance, rates depending rather upon the volume of traffic.[112] The success of the ARPANET has caused considerable attention to be directed to packet switching, and, together with the Pierce Loop, this may well become the keystone of a new networking concept for digital transmission.

INTERFACING

The requirement to interconnect many communications systems in order to create a circuit demands a rational and efficient approach to the problem of interfacing. So far, little significant progress has been made in this direction. The history of analog communications is rife with examples of systems which could not be interconnected due to lack of prior coordination on standards and which then required elaborate redesign or interfacing.[113] One would hope that this situation could be avoided with digital transmission, but there are already many characteristics used in operating

PCM systems (see Fig. 14).[114]

It is not necessary that all systems operate under identical standards, as many of these differences represent design measures to optimize the particular system for the network in which it is used. In fact, too-early standardization may be actually undesirable since it would tend to force the long-haul standards, the requirements for which are not presently known, to conform to the known short-haul requirements.[115] It is desirable, however, to minimize differences that are difficult or impossible to reconcile at the interface.

The digitization of the DCS will require the operation of both PCM and FDM segments for some time, as has been discussed previously. For technical reasons, it is desirable to minimize the interconnections at voice frequency and to perform this at baseband. There are, therefore, three cases to be considered.

1. *PCM-PCM*: Theoretically most advantageous, the direct interconnection of PCM systems at baseband requires great attention to timing and frame alignment. This is further complicated by the absence of a standard hierarchy of digital channel arrangement. Coding problems also arise in this case.

2. *FDM-PCM*: The coding of FDM groups into PCM

Parameter	Value of Parameter	Number of Systems
Sampling rate, kHz	8	All (22)
Number of quantization steps	127	1
	128	19
	239	1
	256	1
Channels per frame	24	18
	30	2
	31	1
	48	1
Time slots per frame	24	15
	25	3
	30	1
	32	2
	48	1
Frames per multiframe	1	4
	3	1
	4	10
	8	3
	6/12	1
	16/32	1
	ϕ	2
Digits per time slot and code at interface	7, SB	1
	8, B	10
	8, SB	10
	10, T	1
Gross digit rate, kHz	1536	9
	1544	6
	1600	3
	1792	1
	1920	1
	2560	1
	3072	1
Companding advantage	18.8	1
	24.1	5
	25	1
	26	8
	26.5	1
	26.8	4
	ϕ	2
Degree of approximation	7 segments or less	5
	9 segments, 3.2:3:3	1
	13 segments	1
	13 segments, slope ratio 2	4
	ϕ segments, slope ratio power of 2	1
	Continuous	7
	ϕ	3
Encoding characteristics	Log, $A = 87.6$	4
	Log, $A = 114$	1
	Log, $\mu = 100$	9
	Log, ϕ	3
	Derived digitally in ternary	1
	Hyperbolic, $\lambda = 20$	1
	ϕ	3
Synchronization method	D	10
	G	5
	Out frame	6
	Bipolar violation	1
Signaling method	First in slot	10
	Time sharing	1
	Eighth in slot	10
	Out slot	1
Load capacity, dBm	0	1
	+2	9
	+3	7
	+4	3
	+5	2

Legend

ϕ—Not yet decided
B—Straight binary
SB—Symmetrical binary
T—Binary coded ternary
D—Distributed
G—Bunched

λ characteristic:

$$y = \frac{(1+\lambda)x}{(1+\lambda x)}$$

A characteristic:

$$y = \frac{Ax}{1 + \log A} \qquad 0 \leqslant v \leqslant V/A$$

$$y = \frac{1 + \log Ax}{1 + \log A} \qquad V/A \leqslant v \leqslant V$$

μ characteristic:

$$y = \frac{\log(1 + \mu x)}{\log(1 + \mu)}$$

Fig. 14.--Fundamental parameters of 22 pulse code modulation systems in various countries. (Franklin & Law, p. 54.)

requires 50 per cent more digital capacity than would be required for the same number of channels coded originally into PCM, but this scheme is feasible in practice.

3. *PCM-FDM*: On present systems, the coding of PCM onto an FDM system wastes much of the dynamic range of the analog system and is believed to be useful only for short-distance applications. Using extremely wideband systems with low SNR, however, this could prove to be a valuable technique.

It can be seen from the above that no problem will arise in making PCM-PCM interconnections within a system provided adequate timing accuracy exists, but serious problems can arise at the interface to another system. Other areas which require close attention in the development of interfacing plans are compatibility of coders and decoders, synchronization, slot changing and frame structure of the PCM signal, and signalling.[117]

The DCA has already established the standards for PCM in the DCS as follows:[118]

1. Sampling rate: 8 kHz.

2. PCM word length: 8 bits, NRZ Code.

3. Signalling: "In-band," utilizing pulse-robbing of the least significant bit of every sixth word.

4. Bit rate: channel, 64 kb/s; gross, 1536 kb/s.

5. Encoding characteristic (best fit): Logarithmic, MU = 255.

6. Channels per frame: 24.

7. Signal-to-quantizing noise ratio: 36 db.

Thus it may be seen from the above that interfacing will be an important requirement in any system which connects to either of the DCS primary interfaces: commercial carriers and tactical systems.[119] Further complicating the specification of these interfaces overseas is the lack of standards on measurement techniques and units of measurement.[120] Much coordination is needed in the area of standards and interface specifications, but it must be done at a level well above the DCA, by international agreement. As it appears unlikely that any sweeping agreements of this sort are soon forthcoming, the interfacing problem is best regarded as a problem to be reckoned with and not one that is capable of being designed out of the system.

SCRAMBLERS

The internal design of many PCM systems is such that repetition of certain pulse sequences can disrupt the system timing. In addition, other sequences may cause power spectra distributions which adversely affect adjacent channels.[121] As it is desirable to furnish the customer with as

nearly transparent an interface to the system as possible, many carriers have incorporated into their systems scrambling devices to cause the line bit stream to be pseudorandom regardless of the user input. Such devices, it is noted, are not unlike cryptographic equipment.

Although the use of scramblers eliminates the problems of nonuniform spectral power distribution, the device, owing to the nature of the randomizing process, is susceptible to generation of multiple output errors upon receipt of a single input error.[122] By extension, it is seen that tandem connection of several scramblers can produce a chain reaction that will generate very many errors at the ultimate destination as a result of only a single error early in the connection.

COST MODELS

One of the more important characteristics of a communications system is the cost of its procurement and operation. Unfortunately, this is also one of the most difficult characteristics to determine accurately. Operating costs can vary widely as a function of location, personnel skill, system loading, and accounting procedures. The model utilized to estimate costs must be valid, a condition difficult to prove or disprove for yet-unbuilt systems.

Finally, if comparisons are to be made between systems, the measurement of costs must be made on a normalized basis that has been derived from an examination of systems designed to perform essentially the same function.

Considerable research into system cost models has revealed several approaches to the problem, all valid within the assumptions of the model. Figure 15 is the model selected for transmission media.[123] It was chosen because the assumptions on which it was based seem reasonable in the DCS context, because it presents comparisons on a normalized basis, and because of its general nature.

The systems used as sources for the results in Figure 15 are wideband (approximately 6 mHz). The curves reflect the impact of both fixed (e.g., procurement) and variable (e.g., operation and maintenance) costs expressed in 1970 dollars. The most significant factor, however, is not the absolute cost reflected but the incremental cost between systems, for it is this that represents the decision variable. In the absence of a technological breakthrough that would significantly reduce the cost of a given system vis-à-vis the others, the relative positions of the curves are unlikely to shift substantially.

Inspection of the data in Figure 15 reveals several things:

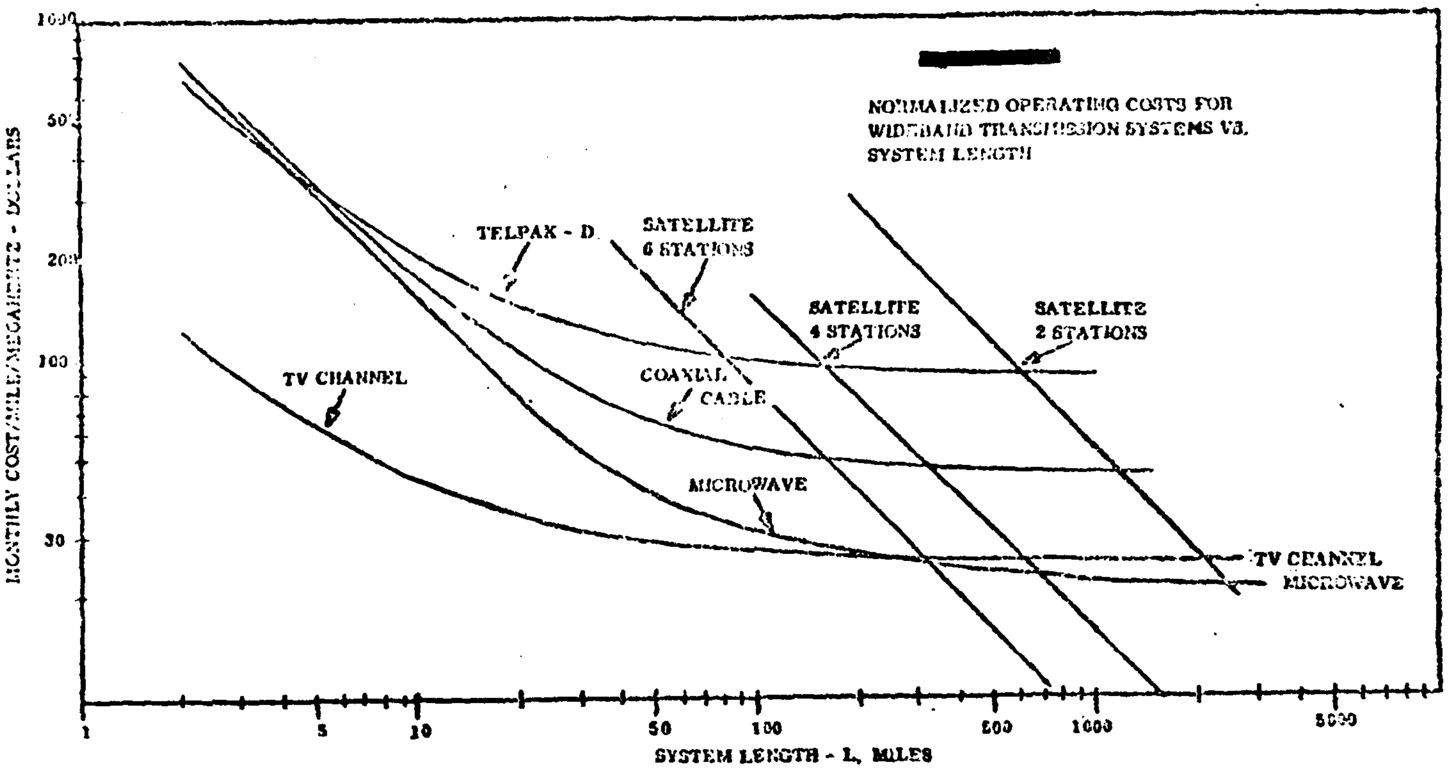


Fig. 15.--System Costs for Various Transmission Media (General Dynamics Electronics Division, p. 4-100.)

1. For systems of 200 miles or less in length, lease of a wideband channel from a common carrier is less costly than construction and operation of an equivalent owned system.

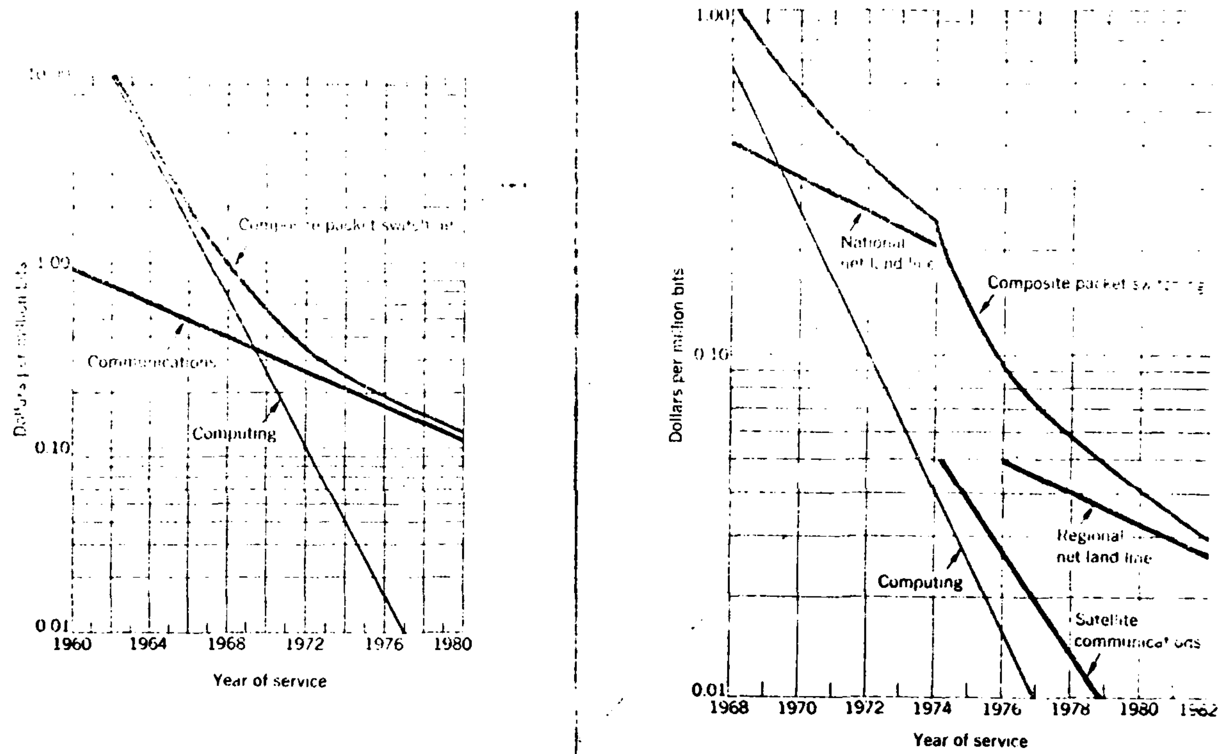
2. Coaxial cable is never less costly than microwave, but for systems of 20 to 30 miles in length the increased stability of that medium may outweigh its slightly increased price.

3. The normalized cost of satellite communications decreases as the number of satellite users increases. As users are added, however, the incremental savings decrease as follows:

User Increase From	To	*Incremental Savings*
2	4	$125/month/mile/mHz
4	6	$26/month/mile/mHz

4. For systems in excess of 200 miles, the use of satellites results in the lowest achievable system cost, especially for multiple use of the satellite.

A communications system is composed of more than merely transmission media. Packet switching depends upon the different rates of cost reduction in computing and communications for its existence.[124] The trends in costs in both these areas are shown in Figure 16. Although the cost



A B

Fig. 16.--Packet-switching costs compared to costs of communications and computing. (Roberts, pp. 50-51.)

of packet switching can never be less than the sum of computing and communications, as computing costs drop rapidly, the cost can approach the pure communications cost quite closely. The curves in Figure 16 consider only the incremental costs of transmitting large quantities of data, not the entire cost of communications. The "B" portion of Figure 16 shows the further decrease expected in packet-switching costs as a result of decreased communications costs resulting from domestic satellite use.

Again, it is the relative positions of the curves of Figure 16 which are most important, not the absolute costs. A general interpretation of these curves is also possible and is expressed in the following extract:

> Except for the numerical absolute scale numbers, it is entirely representative of the costs associated with any system that utilizes both communications and computing components in a fixed ratio! If the communications segment is only a small fraction of the overall system, the crossover will be much later in time, but the identical shape will be preserved. Thus . . . as time passes the costs will become more and more dominated by communications.[125]

The importance of this lies in the need to pursue further system refinements in order to minimize total system cost in either or both of the components of computing and communications.

CHAPTER V

ANALYSIS

INTRODUCTION

The purpose of this chapter is to examine the model of the Defense Communications System (DCS) presented in Chapter II, in the light of the technical and economic considerations presented herein, to determine whether answers to the questions of Chapter I can be deduced from those data. Chapter V also addresses areas which the writer believes are worthy of further investigation in an experimental setting and which have not, to his knowledge, been suggested elsewhere.

As stated at the outset, it is impossible to be exhaustive in any study in this area; there are too much data and knowledge is progressing too rapidly. However, it is essential to examine the goals and objectives of the DCS and then to attempt to determine the best means by which to attain them. In this regard, it is wise to bear in mind a Parkinson's-type Law of Real-Time Systems: "If a communications-based system is sufficiently complex, it is installed before it is designed!"[1] The DCS is in very real danger of

just such a fate unless its aims are firmly stated in the near future.

BENEFITS OF DIGITIZATION

Reasons underlying the move to introduce digital signals into the DCS were stated in Chapter I. In view of the data examined since that point, it can be stated that these reasons, as detailed below, are sound.

1. Increased data handling capability is inherent in a digital-based system, for it has been shown that the efficiency of analog channels operating as digital information carriers is much inferior to the efficiency of digital channels. Furthermore, digital channels can be tailored more easily to the requirements of the data user for traffic volume, either by sub-allocation of a portion of a pulse code modulation (PCM) frame[2] or by allocation of more than one channel. These options do not exist in the analog system, which drives efficiency further downward in that transmission mode.

2. Encryption of voice signals is facilitated by digital transmission, as the PCM process encodes voice into digital form that is acceptable to encryption devices without further modification. As all present encryption devices are digital machines, this is a significant advantage which

should permit widespread use of secure voice in the DCS. However, referring to the model of Figure 2 (page 27), if conversations are to be switched from one node to another utilizing present technology, it is apparent that there are two cases of interest concerning routing. In the first case, routing information is not enciphered but either is used to establish a circuit path for enciphered speech which follows or is added to the enciphered speech according to a prearranged pattern to permit automatic routing through digital nodes. This approach, which amounts to automation of the existing call establishment procedure used in the Automatic Secure Voice System (AUTOSEVOCOM), is technically feasible. The advantage of such an approach is the maintenance of speech security end-to-end; the disadvantage is that traffic flow and routing information are not encrypted and are thus susceptible to being intercepted. In the second case, routing information is enciphered with the speech, which necessitates decryption at each node to extract the data needed to route the call. This approach provides traffic-flow security but does not maintain voice security within the nodes. Lack of internal node voice security could prove to be a serious problem, especially in overseas areas.

Weighing the capabilities of present cryptographic

equipments and the complexity inherent in the equipment required to operate with unenciphered routing data, it appears that the more desirable course of action is the use of enciphered routing information and maintenance of adequate physical security at nodal points. In either case, the elimination of numerous interfaces in the secure speech path which occurs in a digital network greatly improves signal quality and reduces equipment costs.

3. Increased signal quality over longer distances of built-up connections will be provided by a digital system, as shown by the experience of commercial carriers that have adopted the digital system for this reason. Given the DCS standards, it is reasonable to expect that on the 12,000-mile reference circuit an overall signal-to-noise ratio (SNR) of 30 db can be attained on the majority of circuits, especially with a signal-to-quantizing noise ratio of 36 db.

4. A corollary benefit of digitization is lower normalized communications costs. It can be seen from Figure 16 (page 132) that as more of the system can be moved from the communications component into the computing component, a lower normalized communications cost can be realized. Coupled with the inherently greater efficiency of digital systems in handling digital signals, this can

produce significant savings. In fact, one study has concluded that retention of the present analog DCS will cost approximately 20 per cent more than gradual conversion to digital, even excluding the effect of lower operating costs with a digital system.[3]

EFFECT ON THE SYSTEM

An examination of Figure 2 (page 27) reveals that the overall transfer function of any path between several nodes, as path A-D-C-B, depends upon the transfer functions of the links between each pair of nodes. Since these transfer functions are complex, it can be demonstrated mathematically that the overall transfer function of the built-up path is the vector sum of the several link transfer functions. Thus, for the example,

$$\bar{F}_{ADCB}(\omega,t) = \bar{F}_1(\omega,t) + \bar{F}_6(\omega,t) + \bar{F}_4(\omega,t).$$

The importance of this fact arises in the case of tandem connections that involve more than one type of transmission medium. The nature of the transfer function of each type of media has been previously discussed. However, the tandem connection of several of these can result in an extremely complex route transfer function which is beyond the capability of the equalization equipment of the terminating system to accommodate. This is not an unlikely

possibility, as system equalizers are understandably designed to minimize the idiosyncrasies of that system. Thus, even though equalization is not generally utilized in the analog network, it may become necessary to do so at interface points with digital systems, owing to the greater sensitivity of the latter to such impairments as phase distortion. A better solution would be to equalize each link end-to-end. This will very probably be required in an all-digital system, but do do so in an analog system section that is soon to be replaced is not cost-effective on the general scale.

A similar problem arises when hybrid system operation is considered. The feasibility of hybrid operation of digital and analog mode on the same path has been proved by commercial carriers, as discussed. However, the DCS utilizes a greater number of transmission media than the commercial carriers, and their capability to accept hybrid basebands, as has been shown, varies widely. Referring once again to the model (page 27), an interconnection of Link AB (line-of-sight (LOS) microwave) and Link BD (coaxial cable) will pose no serious problem in this regard. However, a connection of Link AB and Link AD (tropospheric scatter) will be virtually impossible in the hybrid mode because Link AD is incapable of operating in that manner. A similar

situation would obtain if Link AD were routed via satellite. Thus, the system designer must choose with great care the routes on which digital service is to be inaugurated, lest entire categories of tandem connections become impossible or at least impracticable. In those cases where such a connection is mandatory, a careful decision must be made as to the method of modulation to be used for the transmission of the "alien" baseband over the intermediary link.

An inspection of Figure 2 (page 27) shows that the failure of a single link need not result in the diminution of traffic between nodes if the links providing alternate paths are capable of handling the traffic that the failed link previously carried. Although this may appear self-evident, it is often lost sight of in the general sense. If the Link AD is a satellite path and Links AB and AC are tropo, a failure of Link AD will require the passage of its traffic via the other two. Being tropo links, however, they may not possess sufficient bandwidth to accept all of this traffic, thereby requiring a reduction in traffic capability. This condition will be exacerbated during the hybrid period of DCS operation for the reasons discussed above, and very careful system planning will be required to minimize this problem.

It has been noted that all switching centers in the

DCS have analog input/output ports. Much ado has been made as to the interfacing requirements this will generate as links are converted to digital mode. However, most of the switching done in the network is at VF, which requires a demultiplexing of the baseband prior to switching whether the link is digital or analog. Thus, the demultiplexer becomes the interface device. In the case of the Automatic Digital Network (AUTODIN), however, the analog interface is a nuisance which restricts data transfer rates to that attainable over voice channels.[4] There is, therefore, a strong temptation to engineer digital systems between AUTODIN switches at an early stage, with little regard for the other traffic on the same route. This may not prove to be the best overall utilization of resources, irrespective of the improvement in AUTODIN efficiency, when one considers that less than 5 per cent of the DCS circuits are AUTODIN access lines and trunks.[5] In general:

> As long as voice traffic predominates, the economics inherent in large cross-sections may reduce the channel cost for data transmission on analog systems to levels well below the costs derived from a small cross-section digital system.[6]

However, as the percentage of digital traffic in the DCS rises, this becomes less and less valid, so the decision as to direct interconnection of AUTODIN centers by digital systems is as much a function of time as of technology.

The advent of an all-digital DCS can be expected to improve significantly the ability of the Defense Communications Agency (DCA) to manage the system. Digital switches are capable of extracting management and status information concurrently with processing traffic and transmitting this to an operations center. At that point, trained personnel could make decisions as to the routing of traffic, the maintenance of areas of the network, etc. In routine instances these decisions could be delegated to a computer, thus permitting true management by exception to be used. Further, accurate system statistics would also be readily available on a near real-time basis, thus permitting both better management and collection of better historical data. Although such an automated management system will be costly, the improvements it would permit could well exceed its cost by several times. In the absence of any meaningful cost data on the DCS, however, it is difficult to evaluate this alternative with any degree of accuracy, and the determination of actual DCS operating costs should be the target of early investigation.

The use of such networks as have been discussed, especially packet-switched networks, makes possible widespread use of adaptive network routing. In adaptive routing, the path to be followed by a given data packet is a

function of the instantaneous status of the network, and the sequential packets of a message may travel to their destination for reassembly by quite varied routes, depending upon the traffic load and link continuity of the system as they transit it. This technique is capable of yielding extremely high reliability, but it demands close control of the network, either by a central authority or by a strict set of rules, else packets can easily orbit the network and magnify local discontinuities. Adaptive routing complements system management by exception, for, if properly programmed, it permits the human manager to concentrate on the truly catastrophic occurrences while routinely taking care of the routine matters. It also permits much higher average utilization of system links since unused links, even if they do not represent the shortest physical route to the destination, are immediately sensed and made available for traffic.

The assumption of increasing DCS data traffic has been accepted throughout this paper as valid. However, the source of that traffic has not been well defined. It is likely, based upon the experience of the civilian sector, that much of this demand originates with users who do not have sufficient volume to warrant a full-time circuit. Under current DCS policy there is no way to obtain data service other than by a dedicated circuit. The variable is

transmission speed; the lower the subscriber volume, the lower the transmission speed authorized, from 150 baud to 2400 baud. It seems beneficial to permit the use of the DCS switched network for the transmission of data by these low-volume users, as is done in the commercial networks. The advent of digital transmission would facilitate such a step for both the system and the user by minimizing the terminal equipment required to interface the equipment to the line.

Because of the unpredictability of the characteristics of a switched circuit assemblage, adaptive equalization is required for data transmission at speeds much above 600 baud. However, the technology of adaptive equalization is now well developed and is available in several commercial modems for use on analog channels. Although it would be possible to institute such service immediately, it is considered unwise owing to the difficulty of controlling the amount of data traffic in the system at any given time and, hence, the system power loading.

Another situation that frequently occurs is a cluster of low-volume users within a relatively small geographical area. The data presented earlier in the paper have stressed the value of maximizing the utilization of the line connecting data sets to computers or to one another, yet

current policy requires a separate line for each user. A measure that could be undertaken at once, without waiting for digitization, is to locate data concentrators and/or multiplexers at the user end of the access line. This would involve allocating only sufficient access lines to handle the peak load and allowing the concentrator to maintain the utilization of that access line at a high level. The result of such an approach would be improved service at lower unit cost.

ECONOMICS

Although system cost, per se, is not likely to be the overriding factor in considering options for the DCS, it cannot be ignored. Especially in recent years, the Congress has demanded detailed economic analysis of alternatives, and there is no reason to expect that to be waived in this instance. Two primary questions come to mind in the subject of costs and conversion to all-digital transmission within the DCS. *First*, are the underlying assumptions that drive the cost estimates valid? *Second*, can the country afford an all-digital DCS?

In general, the cost models presented herein have assumed a constant decrease in the cost of digital components. In the past, this trend has been the case. There is

reason to believe at this time that within the next decade further progress in miniaturization and efficiency of solid state devices will become limited by the laws of physics, not by technology.[7] If this prediction be true, the rate of digital component cost decline will slow markedly and a new balance will be required between communications and computer components of the system. This occurrence would most likely drive down the cost of communications through application of added research effort, with the resultant cost of the system being still reduced in time, albeit at a slower rate than is currently predicted.

The question of affordability is beyond the realm of technical advance; it lies, rather, in the field of politics. The Department of Defense can afford whatever the Congress will authorize it to purchase, not a jot more. For this reason it behooves the planners of the digital DCS to consider the climate into which they must launch their proposal. At the present time the Army has programmed the acquisition of no fewer than five major weapons systems,[8] with programmed expenditures of billions of dollars over the next 10 to 15 years. The Air Force and Navy are in similar circumstances. Given a choice between modernizing weapons systems or communications systems, the Congress is likely to choose the former if history can be a guide to the future.

Indeed, it is difficult to imagine the Secretary of Defense testifying that a new airplane is needed less than a new DCS.

A further complication results from the separation of operational direction of the DCS (which lies with DCA) and implementation responsibility (which lies with the services). Entirely too frequently in the past DCA has not provided sufficient planning data to the services to permit orderly budgeting of communications system additions. The result of such non-coordination is frequently an intraservice competition for resources between the requirements DCA imposes and the needs of that service for its own tactical communications equipment and weaponry. In such a contest the DCS usually finishes second. For as long as the separation of functions persists within the DCS, this writer believes it is certain the services will regard the system as an "outside" competitor for resources and will devote little service initiative toward accomplishment of DCS goals. Such a situation works to the detriment of the system and must be realistically assessed as a factor in any major changes to the system.

Alternatives exist which make the digitization of the DCS more palatable politically than is presently the case, and they should be considered as part of the system

package. This is both politically and economically sound. The following possibilities are offered as a point of departure:

1. Inasmuch as conversion from analog to digital is confidently expected to reduce costs, it should be possible to put the conversion on a "pay-as-you-go" basis after an initial appropriation for equipment. This would require an extended hybrid period and a reduction in the rate of addition of DCS subscribers as compared to the fully-funded phased conversion.

2. Analog equipment no longer required in the DCS can be sold under the Military Assistance Program, with the funds so obtained to be utilized for purchase of new equipment. This can be used either instead of or in conjunction with the scheme above, but it would require enabling legislation.

3. The digital equipment to be used in the DCS can be procured off-the-shelf from a commercial manufacturer in the United States. This would require alteration of the announced DCS digital standards, modification of the equipment, or both; however, the economies of scale should reduce the equipment costs significantly as compared to a militarized system.

4. The DCS can incorporate the remaining federal

communications systems to provide greater communications economy and management efficiency. The present systems often duplicate facilities, as witness the recent General Services Administration (GSA) expenditure of $100 million for a packet-switched data network to satisfy part of the next decade's growth requirement in non-military government communications.[9]

CHAPTER VI

SUMMARY AND RECOMMENDATIONS FOR FUTURE STUDY

SUMMARY

As a result of this study, five major findings, the last with several subparts, have been reached

• Digitization of the DCS will produce the benefits sought, i.e., increased data handling capability, improved secure voice capability, and improved signal quality over longer connections.

• To compensate for the incompatibilities of various media in the hybrid mode, careful system engineering will be required during and after the transition to digital mode to minimize transitional problems.

• Digitization of the network can be expected to facilitate management, especially through use of adaptive network techniques.

• The cost and affordability of a digital network must be considered at an early stage of system design so as to insure that the system designed can be afforded.

• Ten major problem areas are expected to be encountered in conversion from analog to digital mode. They

are as follows:

-- The necessity for hybrid operation over an extended period.

-- Relative unfamiliarity of operating personnel with digital techniques. This will demand increased training and may require a major adjustment in personnel assignment policies to support the system during the hybrid stage, when technicians who possess both digital and analog skills will be required.

-- Definition of system performance criteria in a meaningful form for digital and hybrid networks and for varying types of source signals.

-- Interfacing with common carriers and other military networks which operate by using technical standards that are different from those in the DCS.

-- Resolution of the rates charged by common carriers for data service, especially for short duration connected service over lines not dedicated to the DCS.

-- The distribution of government communications among the many data common carriers will have significant political and economic impact as the portion of the DCS using digital transmission expands.

-- Choice of timing system and its interface with different timing methods.

-- Accurate and effective system performance monitoring and quality control.

-- Early determination of system routing and switching philosophy.

-- System management doctrine and organization, to include accurate analysis and prediction of traffic trends by type of traffic and source.

TOPICS FOR FUTURE STUDY

Review and analysis of the data presented have yielded several areas which this writer believes deserve further study. In particular, the two topics below do not appear to have been examined elsewhere.

- The Advanced Research Projects Agency Net (ARPA-NET) of the Department of Defense was seen to have a typical network transit of 100 msec for a data packet. *Comité Consultatif International Télégraphique et Téléphonique* (CCITT) standards for round-trip circuit delay state that less than 300 msec total delay is acceptable without reservation on a voice circuit.[1] There seems to be no theoretical reason digitized speech cannot be transmitted on a packet-switched network with acceptable delay. Extremely reliable transmission should be possible by utilizing adaptive network techniques. The use of concentrators or compressive storage

will compensate for any additional transit speed that might be required to attain an acceptable delay figure overall in a network larger and more complex than ARPANET.

• The suitability of packet-switching to adaptive routing is attractive in military communications networks, as is the simplicity obtainable with Pierce Loops. The delay associated with a Pierce Loop is a function of the circumference of the loop and can become excessive on very long loops. It appears that Pierce Loops are suitable for local plant interconnections, to include digitized voice if suitable control features are provided to control delay and sequence packets properly. These loops can then be interconnected by means of a multinode packet-switching network to provide rapid, low-cost trunking with the capability of adaptive routing. A conceptual diagram of such a network is shown in Figure 17.

The separation of operational direction and implementation responsibility within the DCS has been a source of conflict since formation of the system. This writer believes it is imperative to examine thoroughly the management structure of the DCS to determine whether it can be altered to improve system responsiveness, decrease the amount of detailed centralized management, and reduce redundancy throughout the system.

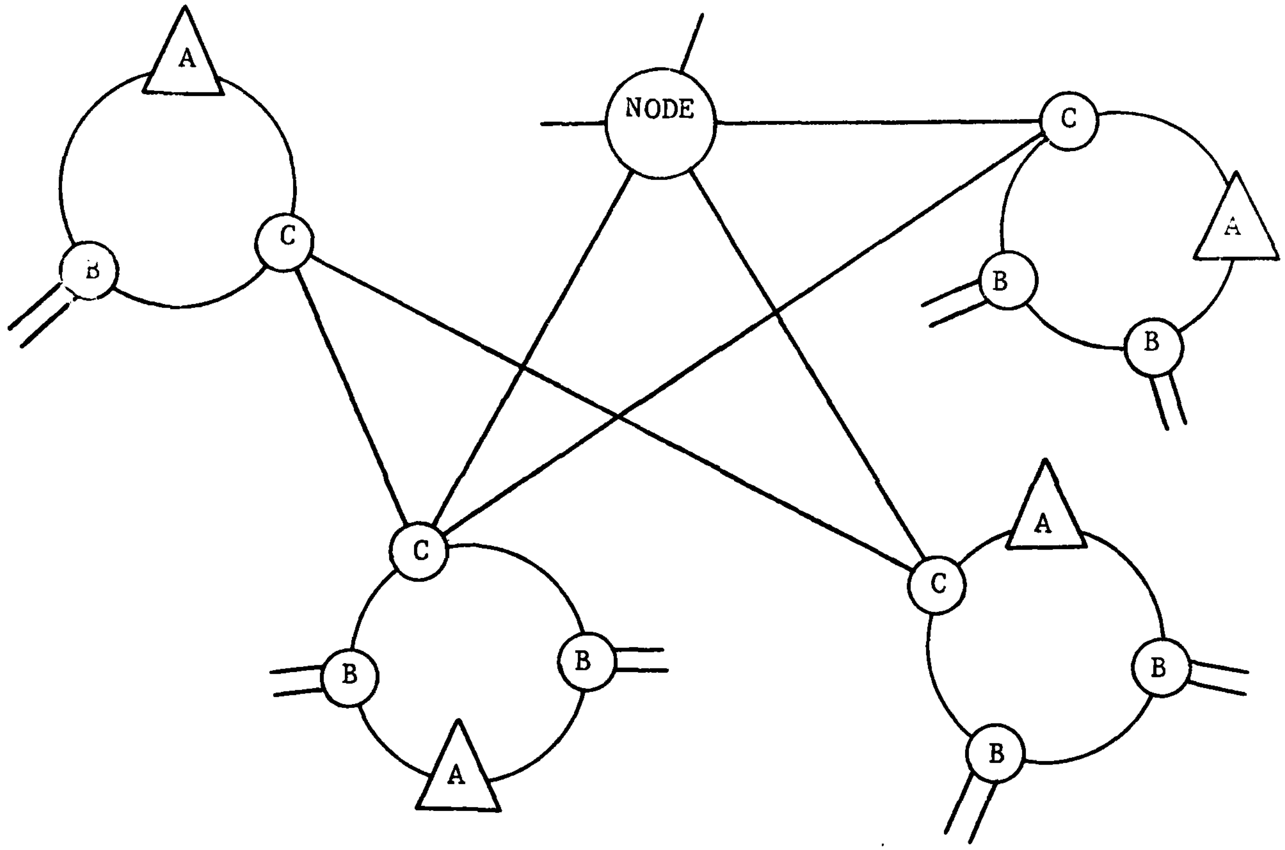


Fig. 17.--Composite Pierce Loop--Packet Switched Network

Redesign of any communications system is necessarily dependent upon accurate prediction of service requirements. It has been shown that current growth estimates within the DCS vary by factors of three or four, which makes system planning difficult at best. Development of a system within the DCS to accurately monitor past and present service demands and traffic volumes is urgently required so these data may be compared with future system plans to provide meaningful engineering goals.

On the principle that "the more communications you have, the more you want," expansion of the DCS in the past has often created demands for yet more extensive service. This will probably continue, especially as services like secure voice and remote computation are offered, because the user perceives the DCS as a "free" service. This writer recommends that serious consideration be given to the relative merits of placing the DCS on an industrial funding basis to minimize uneconomical demands on the system.

The conversion of the DCS to digital form appears to be an idea whose time has come. It is imperative, however, that a change of such proportions be carefully planned and executed, lest the remedy prove worse than the problem it was created to solve.

NOTES

NOTES

CHAPTER I

[1]A. A. Childers, "The Introduction of Digital Transmission into the DCS," *Signal*, XXV, No. 7 (March 1971), 42.

[2]R. P. Witt, S. Barosin, and R. A. Culpon, *An Economic Comparison of Analog vs. Digital-Based Common User Communications*, MTR-1930, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, November 1970), p. 55.

[3]G. T. Gould, Jr., "Military Communications in the Information Age," *Signal*, XXV, No. 1 (September 1970), 12.

[4]R. P. Witt, S. Barosin, and K. D. Kasparian, *Air Force Planning for Digital Communications*, MTR-1803, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, August 1970), p. 1.

[5]U.S., Congress, House, Committee on Armed Services, *Review of Department of Defense Worldwide Communications, Phase I* [*Mollohan Report*], Armed Services Investigating Subcommittee report, 92d Cong., 1st sess., 10 May 1971, p. 4.

[6]Childers, p. 42; and Witt, Barosin, and Kasparian, p. 4.

[7]Howard Falk, "Technology '74: Data Communications," *IEEE Spectrum*, XI, No. 1 (January 1974), 36; and R. A. Peters and K. M. Simpson, "Eurodata: Data Communications in Europe, 1972-1985," *Datamation*, December 1973, pp. 77-80.

[8]K. E. Fultz and D. B. Penick, "The T1 Carrier System," *Bell System Technical Journal*, XLIV, No. 7 (September 1965), 1405.

[9]Howard Falk, "Technology '74: Communications and Microwaves," *IEEE Spectrum*, XI, No. 1 (January 1974), 35.

CHAPTER II

[1]H. A. Schulke, Jr., "The DCS Today and How It Operates," *Signal*, XXV, No. 11 (July 1971), 22; see also: T. O. Duff, "The Defense Communications Agency" (U.S. Air Force Air War College Report #4564, April 1972), pp. 14-15.

[2]Schulke, p. 22.

[3]L. K. Wentz, *Overseas AUTOVON Anomalies*, MTR-2761 (Boston: MITRE Corporation, 8 January 1974), p. 1.

[4]Schulke, p. 23.

[5]U.S., Congress, House, Committee on Armed Services, *Review of Department of Defense Worldwide Communications, Phase I* [*Mollohan Report*], Armed Services Investigating Subcommittee report, 92d Cong., 1st sess., 10 May 1971, p. 23.

[6]Wentz, p. 1.

[7]Schulke, p. 23; and personal experience.

[8]Schulke, p. 23.

[9]G. T. Gould, Jr., "Military Communications in the Information Age," *Signal*, XXV, No. 1 (September 1970), 12.

[10]F. L. Mills, "Centralization of the U.S. Army Telecommunications Computer Programming Effort" (U.S. Army Command and General Staff College MMAS thesis, 1971), p. 22.

[11]Gould, p. 12.

[12]R. P. Witt, S. Barosin, and K. D. Kasparian, *Air Force Planning for Digital Communications*MTR-1803, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, August 1970), p. 5.

[13]R. P. Witt, S. Barosin, and R. A. Culpon, *An Economic Comparison of Analog vs. Digital-Based Common User Communications*, MTR-1930, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, November 1970), pp. 9-19; see also: P. E. Law, "Communications System Planning," *Signal*, XXVII, No. 4 (December 1972), 22.

[14] Schulke, p. 22.

[15] Schulke, p. 22; and J. H. Weiner, "U.S. Military Worldwide Communications," *IEEE Spectrum*, III, No. 10 (October 1966), 80.

[16] J. E. Kessler, "The Transmission Performance of Bell System Toll Connecting Trunks," *Bell System Technical Journal*, L, No. 8 (October 1971), 2742-45; see also: "The Difficult Problem of Exchange Trunk Carrier," *Lenkurt Demodulator*, October 1960, p. 2.

[17] Defense Communications Agency, *System Description: DCS-AUTODIN*, DCA Circular 370-D95-1 (November 1973), p. 3-1; and Schulke, p. 22.

[18] General Dynamics Electronics Division, *Study of Electronic Handling of Mail: Transmission Methods*, Report R-69-046-5, Prepared for Bureau of Research and Engineering, Post Office Department, Washington (July 1970), p. 1-2; see also: Witt, Barosin, and Kasparian, pp. 7-8.

[19] Duff, p. 40.

[20] J. P. Molnar, "A Telecommunications Superhighway for the 1970's," *Signal*, XXV, No. 1 (September 1970), 47.

[21] *Ibid.* [22] U.S., Congress, House, pp. 2-4.

[23] Could, p. 12.

[24] Witt, Barosin, and Culpon, p. 30.

[25] Witt, Barosin, and Culpon, p. 5. [26] Duff, p. 38.

[27] According to data furnished by the U.S. Army Command and General Staff College Air Force Section, cost of the C5 fleet is taken as $2.84 billion (79 aircraft at flyaway cost of $36 million each).

[28] A. A. Childers, "The Introduction of Digital Transmission into the DCS," *Signal*, XXV, No. 7 (March 1971), 42; and Witt, Barosin, and Culpon, p. 55.

[29] Mills, p. 22.

[30]J. Martin, *Systems Analysis for Data Transmission* (Englewood Cliffs, N. J.: Prentice-Hall, 1972), p. 261.

[31]This was confirmed by personal sources at Headquarters, Defense Communications Agency.

[32]Defense Communications Agency, p. 8-2; and Janet L. Norman, "Reducing Telephone Network Errors," *Datamation*, 1 October 1971, pp. 24-25.

[33]J. R. Pierce, "Some Practical Aspects of Digital Transmission," *IEEE Spectrum*, V, No. 11 (November 1968), 68-70.

[34]*Ibid.*, pp. 69-70. [35]Childers, p. 42.

[36]Based upon figures from Defense Communications Agency, p. 3-8, and Schulke, p. 22.

CHAPTER III

[1]"The Development of Multiplex Telephony," *Lenkurt Demodulator*, XIV, No. 12 (December 1965), 5.

[2]M. D. Balkovic *et al.*, "High-Speed Voiceband Data Transmission Performance on the Switched Telecommunications Network," *Bell System Technical Journal*, L, No. 4 (April 1971), 1350.

[3]J. E. Kessler, "The Transmission Performance of Bell System Toll Connecting Trunks," *Bell System Technical Journal*, L, No. 8 (October 1971), 2742.

[4]W. R. Bennett and J. R. Davey, *Data Transmission* (New York: McGraw-Hill, 1965), p. 29.

[5]*Ibid.*, pp. 34-35.

[6]"Multiplexing and Modulation in Carrier Telephone Systems" [hereinafter referred to as "Multiplexing and Modulation"], *Lenkurt Demodulator*, VIII, No. 1 (January 1959), 4.

[7]Bennett and Davey, p. 36.

[8]Bennett and Davey, p. 39.

[9]"Multiplexing and Modulation," p. 4.

[10]"Methods for Transmitting Data Faster" [hereinafter referred to as "Transmitting Data Faster"], *Lenkurt Demodulator*, IX, No. 3 (March 1960), 5.

[11]*Ibid.*, p. 6.

[12]"Requirements of a Human Communications Channel" [hereinafter referred to as "Human Communications Channel"], *Lenkurt Demodulator*, September 1965, p. 4.

[13]"Load Capacity of High-Density Multiplex Systems" [hereinafter referred to as "Load Capacity"], *Lenkurt Demodulator*, XIV, No. 3 (March 1965), 6-8.

[14]H. Rudin, Jr., "Data Transmission: A Direction for Future Development," *IEEE Spectrum*, VII, No. 2 (February 1970), 81.

[15]"Phase Jitter," *GTE Lenkurt Demodulator*, December 1972, p. 2.

[16]"Load Capacity," pp. 6-7.

[17]"A Glossary of Signaling Terms," *GTE Lenkurt Demodulator*, April 1974, pp. 6-7.

[18]D. H. Hamsher, *Communication System Engineering Handbook* (New York: McGraw-Hill, 1967), p. 3-5.

[19]"Load Capacity," p. 10. [20]Kessler, p. 2768.

[21]Bennett and Davey, pp. 84-85.

[22]"Human Communications Channel," p. 7.

[23]Kessler, p. 2766.

[24]"The Theory and Use of Compandors in Voice Transmission Systems" [hereinafter referred to as "Compandors"], *Lenkurt Demodulator*, October 1964, p. 11.

[25]R. G. Gould and G. K. Helder, "Transmission Delay and Echo Suppression," *IEEE Spectrum*, VII, No. 4 (April 1970), 50-53.

[26]"Human Communications Channel," p. 4.

[27]Paul Baran, "LSI: The Basic Module for New Communication Networks" (paper presented before IEEE Computer Group Conference on LSI, Los Angeles, 27-28 June 1968), pp. 1-6.

[28]H. Kahl, "Evolution of Wideband Data Services," *Bell Laboratories Record*, XLIX, No. 8 (September 1971), 244-50.

[29]E. Brenner and M. Javid, *Analysis of Electric Circuits* (New York: McGraw-Hill, 1959), pp. 623-72.

[30]"Transmitting Data Faster," p. 3.

[31]Bennett and Davey, pp. 30-31.

[32]C. E. Shannon, "A Mathematical Theory of Communication," *Bell System Technical Journal*, XXVII: No. 6 (July 1948), 379-423; and No. 8 (October 1948), 623-56.

[33]J. W. Bayless, S. J. Campanella, and A. J. Goldberg, "Voice Signals: Bit-by-Bit," *IEEE Spectrum*, X, No. 10 (October 1973), 28-29.

[34]H. Nyquist, "Certain Topics in Telegraph Transmission Theory," *Transactions of the AIEE*, XLVII, No. 4 (April 1928), 617-44.

[35]R. H. Franklin and H. B. Law, "Trends in Digital Communications by Wire," *IEEE Spectrum*, III, No. 11 (November 1966), 54.

[36]Bayless, Campanella, and Goldberg, p. 29.

[37]Bayless, Campanella, and Goldberg, pp. 28-29 & 34.

[38]Bayless, Campanella, and Goldberg, pp. 29-30.

[39]Bayless, Campanella, and Goldberg, p. 29; and A. A. Childers, "The Introduction of Digital Transmission into the DCS," *Signal*, XXV, No. 7 (March 1971), 43.

[40]Baran, pp. 11-13.

[41]Bayless, Campanella, and Goldberg, pp. 29-31.

[42]Howard Falk, "Technology '74: Communications and Microwaves," *IEEE Spectrum*, XI, No. 1 (January 1974), 35.

[43]Bayless, Campanella, and Goldberg, pp. 28 & 31-34.

[44]Bayless, Campanella, and Goldberg, p. 34.

[45]Defense Communications Agency, *System Description: DCS-AUTODIN*, DCA Circular 370-D95-1 (November 1973), p. 2-6.

[46]B. Glance, "Power Spectra of Multilevel Digital Phase-Modulated Signals," *Bell System Technical Journal*, L, No. 9 (November 1971), 2857-78; see also: "Load Capacity," pp. 9-10.

[47]Rudin, pp. 79-82. [48]Rudin, p. 81.

[49]E. Y. Ho and D. A. Spaulding, "Data Transmission Performance in the Presence of Carrier Phase Jitter and Gaussian Noise," *Bell System Technical Journal*, LI, No. 8 (October 1972), 1927-31.

[50]P. B. Cunningham *et al.*, "A New T-1 Regenerative Repeater," *Bell Laboratories Record*, XLIX, No. 4 (April 1971), 109-114.

[51]"PCM Field Study," *GTE Lenkurt Demodulator*, March 1974, p. 2.

[52]Bennett and Davey, pp. 94 & 291-303.

[53]Bennett and Davey, p. 94.

[54]Brenner and Javid, p. 667.

[55]Bennett and Davey, pp. 83-90.

[56]H. Rudin, Jr., "Automatic Equalization Using Transversal Filters," *IEEE Spectrum*, IV, No. 1 (January 1967), 53-59; R. W. Lucky, "Automatic Equalization for Data Communication," *Bell System Technical Journal*, XLIV, No. 4 (April 1965), 547-88; and R. W. Lucky and H. R. Rudin, "An Automatic Equalizer for General-Purpose Communications Channels," *Bell System Technical Journal*, XLVI, No. 9 (November 1967), 2179-2208.

[57]Kessler, p. 2772.

[58]Bennett and Davey, pp. 90-91.

[59]"Compandors," p. 11.

[60]Gould and Helder, pp. 47-54.

[61]Personal experience with such systems.

[62]Bennett and Davey, pp. 91-92.

[63]Bennett and Davey, pp. 92 & 94.

[64]Ho and Spaulding, pp. 1927-31; and "Phase Jitter," pp. 2-7.

[65]J. S. Mayo, "Experimental 224 Mb/s PCM Terminals," *Bell System Technical Journal*, XLIV, No. 9 (November 1965), 1828-41; see also: "PCM Field Study," pp. 3-4.

[66]Mayo, pp. 1828-41.

[67]J. H. Fennick, "Transmission Objectives for Voice-band Data," *Bell Laboratories Record*, L, No. 1 (January 1972), 18-21.

[68]Mayo, pp. 1813-41. [69]Fennick, p. 18.

[70]R. P. Witt, S. Barosin, and K. D. Kasparian, *Air Force Planning for Digital Communications*, MTR-1803, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, August 1970), p. 4.

[71]Baran, pp. 2 & 25. [72]Baran, p. 2.

[73]Witt, Barosin, and Kasparian, p. 3.

[74]Bayless, Campanella, and Goldberg, pp. 28-36.

CHAPTER IV

[1]C. S. Berger, "The Old TD2 Becomes the New TD2C," *Bell Laboratories Record*, LI, No. 9 (October 1973), 279-84.

[2]General Dynamics Electronics Division, *Study of Electronic Handling of Mail: Transmission Methods*, Report R-69-046-5, Prepared for Bureau of Research and

Engineering, Post Office Department, Washington (July 1970), p. 3-5.

[3] *Ibid.*, p. 3-6. [4] *Ibid.*, p. 3-7.

[5] J. W. Bayless, S. J. Campanella, and A. J. Goldberg, "Voice Signals: Bit-by-Bit," *IEEE Spectrum*, X, No. 10 (October 1973), 28.

[6] General Dynamics Electronics Division, p. 3-6; and M. J. Thompson, "Transmitting Digital Signals on Analog Radio Links," *Bell Laboratories Record*, XLIX, No. 4 (April 1971), 102-108.

[7] Thompson, pp. 102-108.

[8] J. E. Kessler, "The Transmission Performance of Bell System Toll Connecting Trunks," *Bell System Technical Journal*, L, No. 8 (October 1971), 2746.

[9] J. H. Weiner, "U.S. Military Worldwide Communications," *IEEE Spectrum*, III, No. 10 (October 1966), 80.

[10] Paul Baran, "LSI: The Basic Module for New Communication Networks" (paper presented before IEEE Computer Group Conference on LSI, Los Angeles, 27-28 June 1968), p. 9.

[11] *Ibid.*, p. 11.

[12] "PCM-FDM Compatibility," *GTE Lenkurt Demodulator*, July 1971, p. 2.

[13] "PCM Cable Considerations," *GTE Lenkurt Demodulator*, June 1973, p. 2.

[14] *Ibid.*, p. 6.

[15] G. H. Burwell *et al.*, "A Portable Test Set for Detecting Errors on T-1 Lines," *Bell Laboratories Record*, L, No. 1 (January 1972), 14-17.

[16] *Ibid.*; and P. B. Cunningham *et al.*, "A New T-1 Regenerative Repeater," *Bell Laboratories Record*, XLIX, No. 4 (April 1971), 109-114.

[17]J. J. Karakash, *Transmission Lines and Filter Networks* (New York: Macmillan, 1950), pp. 25-28.

[18]General Dynamics Electronics Division, p. 3-22.

[19]J. S. Mayo, "Experimental 224 Mb/s PCM Terminals," *Bell System Technical Journal*, XLIV, No. 9 (November 1965), 1813-41.

[20]J. F. Gunn *et al.*, "Mastergroup Digital Transmission on Modern Coaxial Systems," *Bell System Technical Journal*, L, No. 2 (February 1971), 501-520.

[21]A. A. Childers, "The Introduction of Digital Transmission into the DCS," *Signal*, XXV, No. 7 (March 1971), 42.

[22]General Dynamics Electronics Division, p. 3-36.

[23]Personal experience in Southeast Asia, 1969.

[24]Personal experience in Southeast Asia, 1969.

[25]General Dynamics Electronics Division, p. 3-36.

[26]*COMSAT Annual Report to the President and the Congress, 1973* [hereinafter referred to as *COMSAT Annual Report*] (Washington, 31 July 1973), p. 21.

[27]U.S. Army Signal School, *Satellite Communications*, Special correspondence course (1968), pp. 4-1 to 4-8.

[28]*Ibid.* [29]*COMSAT Annual Report*, p. 50.

[30]Personal experience. [31]Childers, p. 42.

[32]Data received informally from personal sources within the Defense Communications Agency.

[33]"Data Timing for HF Radio Transmission," *Lenkurt Demodulator*, February 1964, pp. 2-3.

[34]*Ibid.*, pp. 3-5.

[35]U.S. Army Electronics Command, *Suitability of Existing Army Communications for Data Transmission*, Technical Report ECOM-2801, By J. R. Tucker and A. H. Gillette

(Fort Monmouth, N. J., January 1967), p. 31; and personal experience.

[36] *All-Digital 1980 DCS Study, Phase I, Task IV A, Final Report (U)* [hereinafter referred to as *All-Digital 1980 DCS Study (U)*], Prepared by IBM and Philco-Ford under Contract DCA 100-70-C-0001 (November 1970), p. 3-12.

[37] Information received during conversations with Squadron Leader John Jamieson, Royal Air Force, at Supreme Headquarters Allied Powers Europe, 1970-1973.

[38] *All-Digital 1980 DCS Study (U)*, p. 4-93.

[39] R. P. Witt, S. Barosin, and R. A. Culpon, *An Economic Comparison of Analog vs. Digital-Based Common User Communications*, MTR-1930, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, November 1970), p. 18.

[40] *All-Digital 1980 DCS Study (U)*, p. 4-94.

[41] *All-Digital 1980 DCS Study (U)*, pp. 4-93 & 4-100.

[42] J. P. Molnar, "A Telecommunications Superhighway for the 1970's," *Signal*, XXV, No. 1 (September 1970), 44-47.

[43] E. E. Altshuler, V. J. Falcone, and K. N. Wulfsberg, "Atmospheric Effects on Propagation at Millimeter Wavelengths," *IEEE Spectrum*, V, No. 7 (July 1968), 83-90.

[44] Molnar, p. 45.

[45] J. P. Molnar, "The Telephone Plant of the 1970's," *Bell Laboratories Record*, XLIX, No. 1 (January 1971), 2-12.

[46] R. Bowers and J. Frey, "Technology Assessment and Microwave Diodes," *Scientific American*, CCXXVI, No. 2 (February 1972), 15.

[47] Molnar, "Telecommunications Superhighway," p. 46.

[48] "Millimeter Waveguide To Be Tested in West Germany," *Electronics*, XLVII, No. 4 (21 February 1974), 55.

[49] Molnar, "Telecommunications Superhighway," p. 47.

[50] G. White, "A One-Gigabit-Per-Second Optical PCM

Communications System," *Proceedings of the IEEE*, LVIII, No. 10 (October 1970), 1779-80; see also: G. White, "Optical Modulation at High Information Rates," *Bell System Technical Journal*, L, No. 8 (October 1971), 2607-2645.

[51]Molnar, "Telecommunications Superhighway," p. 47.

[52]R. A. Peters and K. M. Simpson, "Eurodata: Data Communications in Europe, 1972-1985," *Datamation*, December 1973, pp. 76-80.

[53]S. L. Mathison and P. M. Walker, "Regulatory and Economic Issues in Computer Communications," *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1261.

[54]*Digital Data System Channel Interface Specifications (Preliminary)*, Bell System Publication 41021 (March 1973), p. 1.

[55]*Ibid.*, p. 3.

[56]H. C. Fleming and R. M. Hutchinson, Jr., "Low-Speed Data Transmission Performance on the Switched Telecommunications Network," *Bell System Technical Journal*, L, No. 4 (April 1971), 1385-1405; see also: Kessler, pp. 2741-78.

[57]J. Martin, *Systems Analysis for Data Transmission* (Englewood Cliffs, N. J.: Prentice-Hall, 1972), pp. 245-59.

[58]"Keep an Eye on Bell's ISAL [Information System Access Lines] Tariff," *Business Automation* (News Analysis Edition), 15 February 1971, pp. 6-7 & 12.

[59]J. H. Bobsin and L. E. Forman, "The T2 Digital Line--Extending the Digital Network," *Bell Laboratories Record*, LI, No. 8 (September 1973), 239.

[60]Cunningham *et al.*, pp. 109-114.

[61]I. Welber, "A Commitment," *Bell Laboratories Record*, L, No. 1 (January 1972), inside front cover.

[62]Bobsin and Forman, p. 239.

[63]K. E. Fultz and D. B. Penick, "The T1 Carrier

System," *Bell System Technical Journal*, XLIV, No. 7 (September 1965), 1405-1451.

[64]R. T. James and P. E. Muench, "AT&T Facilities and Services," *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1345.

[65]*Ibid.*, pp. 1347-48.

[66]"9101C PCM Repeatered Line Equipment," *GTE Lenkurt Practices*, Issue 2 (November 1972), sec. 342-910-103, p. 3.

[67]J. E. Cox, "Western Union Digital Services," *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1352.

[68]*Ibid.*, p. 1353.

[69]Allen R. Worley, "The DATRAN System," *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1357-58.

[70]"AT&T's Rate Plans Jolt the Competition," *Business Week*, 3 March 1973, p. 26.

[71]U.S. Army Electronics Command, *Performance Rating for Analogue and Digital Modulation Methods*, Research and Development Technical Report ECOM-3269, By F. S. Gutleber (April 1970), pp. 1-13.

[72]W. R. Bennett and J. R. Davey, *Data Transmission* (New York: McGraw-Hill, 1965), p. 238.

[73]"Concentrators and Multiplexers," *GTE Lenkurt Demodulator*, August 1973, p. 1.

[74]H. Rudin, Jr., "Data Transmission: A Direction for Future Development," *IEEE Spectrum*, VII, No. 2 (February 1970), 81-82.

[75]"Concentrators and Multiplexers," pp. 2-7.

[76]N. Abramson, *The Aloha System--Another Alternative for Computer Communications*, University of Hawaii Technical Report B70-1 (April 1970), p. 14.

[77]Gerald Lapidus, "New Solid-State Circuits," *IEEE Spectrum*, XI, No. 1 (January 1974), 99.

[78]"Subscriber Carrier," *Lenkurt Demodulator*, March 1970, p. 2.

[79]Witt, Barosin, and Culpon, pp. 26-28.

[80]Mayo, p. 1822.

[81]Mayo, pp. 1822-23.

[82]Mayo, pp. 1823-25.

[83]Mayo, pp. 1825-27.

[84]R. G. McKay, "Digital Communications--A Tutorial," *Bell Laboratories Record*, XLIX, No. 9 (October 1971), 283.

[85]U.S. Army Test & Evaluation Command, *Data Transmission Equipment*, MTP 6-2-065 (25 September 1969), p. A-1.

[86]*All-Digital 1980 DCS Study (U)*, p. 3-11.

[87]*All-Digital 1980 DCS Study (U)*, pp. 3-11 & 3-12.

[88]J. C. McPherson, "Data Communication Requirements of Computer Systems," *IEEE Spectrum*, IV, No. 12 (December 1967), 44.

[89]B. Combs, "Tymnet: A Distributed Network," *Datamation*, July 1973, pp. 40-41.

[90]J. H. Fennick, "Transmission Objectives for Voiceband Data," *Bell Laboratories Record*, L, No. 1 (January 1972), 18.

[91]*Ibid.*, pp. 18-21.

[92]Kessler, p. 2758; see also: Kessler, p. 2774.

[93]Fleming and Hutchinson, pp. 1370-73; and Janet L. Norman, "Reducing Telephone Network Errors," *Datamation*, 1 October 1971, pp. 26-30.

[94]Norman, p. 26; see also: Fleming and Hutchinson, p. 1373.

[95]Norman, p. 26.

[96]R. W. Blair and R. S. Burnell, "Monitors Take the Pulse of T-1 Transmission Lines," *Bell Laboratories Record*, LI, No. 2 (February 1973), 55-60.

[97] Worley, pp. 1360-61; and information received directly from Defense Communications Agency sources who had toured the DATRAN Vienna, Va., control site.

[98] E. O. Elliott, "A Model of the Switched Telephone Network for Data Communications," *Bell System Technical Journal*, XLIV, No. 1 (January 1965), 89-109.

[99] Defense Communications Agency, *Operational Direction and Status Reporting for the Defense Communications System*, DCA Circular 310-55-1 (July 1972), vol. ii.

[100] Burwell *et al.*, pp. 14-17.

[101] "PCM Cable Considerations," pp. 2-4.

[102] W. H. C. Higgins, "Major Trends in Switching," *Bell Laboratories Record*, L, No. 9 (October 1972), 272-73.

[103] Witt, Barosin, and Culpon, p. 5.

[104] G. D. Johnson, "No. 4 ESS--Long Distance Switching for the Future," *Bell Laboratories Record*, LI, No. 8 (September 1973), 226.

[105] Higgins, p. 276.

[106] J. R. Pierce, "Network for Block Switching of Data," *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1133-46.

[107] C. H. Coker, "An Experimental Interconnection of Computers Through a Loop Transmission System," *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1175.

[108] W. J. Kropfl, "An Experimental Data Block Switching System," *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1147-66.

[109] B. Avi-Itzhak, "Heavy Traffic Characteristics of a Circular Data Network," *Bell System Technical Journal*, L, No. 8 (October 1971), 2521-50.

[110] L. G. Roberts, "Data by the Packet," *IEEE Spectrum*, XI, No. 2 (February 1974), 48.

[111] *Ibid.*, p. 46. [112] *Ibid.*, pp. 47-50.

[113]R. H. Franklin, "Compatibility in World Communications," *IEEE Spectrum*, III, No. 10 (October 1966), 73-77.

[114]R. H. Franklin and H. B. Law, "Trends in Digital Communications by Wire," *IEEE Spectrum*, III, No. 11 (November 1966), 54.

[115]*Ibid.*, p. 59. [116]*Ibid.*, p. 55.

[117]*Ibid.*, pp. 57-59. [118]Childers, pp. 42-43

[119]*All-Digital 1980 DCS Study (U)*, p. 6-1.

[120]Franklin, pp. 73-77.

[121]H. Kahl, "Evolution of Wideband Data Services," *Bell Laboratories Record*, XLIX, No. 8 (September 1971), 250.

[122]M. D. Balkovic *et al.*, "High-Speed Voiceband Data Transmission Performance on the Switched Telecommunications Network," *Bell System Technical Journal*, L, No. 4 (April 1971), 1357-58.

[123]General Dynamics Electronics Division, p. 4-100.

[124]Roberts, pp. 50-51. [125]Roberts, p. 50.

CHAPTER V

[1]J. Martin, *Systems Analysis for Data Transmission* (Englewood Cliffs, N. J.: Prentice-Hall, 1972), p. 376.

[2]"U.S. Markets, 1974: Communications," *Electronics*, XLVII, No. 1 (10 January 1974), 108.

[3]R. P. Witt, S. Barosin, and R. A. Culpon, *An Economic Comparison of Analog vs. Digital-Based Common User Communications*, MTR-1930, Contract F19(628)-68-C-0365 (Bedford, Mass.: MITRE Corporation, November 1970), p. 54.

[4]Defense Communications Agency, *System Description: DCS-AUTODIN*, DCA Circular 370-D95-1 (November 1973), pp. 6-1 to 6-7.

[5]Based upon figures taken from H. A. Schulke, Jr., "The DCS Today and How It Operates," *Signal*, XXV, No. 11

(July 1971), 22, and Defense Communications Agency, p. 3-8.

[6]R. G. McKay, "Digital Communications--A Tutorial," *Bell Laboratories Record*, XLIX, No. 9 (October 1971), 284.

[7]Gerald Lapidus, "New Solid-State Circuits," *IEEE Spectrum*, XI, No. 1 (January 1974), 99.

[8]Specifically, these are the main battle tank (XM-1), mechanized infantry combat vehicle (MICV), advanced assault helicopter (AAH), utility tactical transport aircraft system (UTTAS), and the surface-to-air missile, developmental (SAM-D).

[9]"The Largest Civilian Government Communications Procurement in History," *Data Channels*, I, No. 2 (March 1974), 4-5.

CHAPTER VI

[1]R. G. Gould and G. K. Helder, "Transmission Delay and Echo Suppression," *IEEE Spectrum*, VII, No. 4 (April 1970), 54.

GLOSSARY

GLOSSARY

Selected terms used in this paper are defined as follows. Standard usage is incorporated wherever possible, but in the event of conflict the definition below takes precedence in this work.

Analog signal: an electrical signal that varies continuously in time with no discontinuities. Mathematically, a continuous function of time.

Analog system *or* Analog communication system: a communications system which conveys information by the variation of the magnitude, phase, or frequency of an analog signal, and which accomplishes the routing of these information-bearing signals.

ARPANET: Advanced Research Projects Agency Net; a Department of Defense computer communications network.

ARQ: Automatic Request for Retransmission; a mode of operation of data terminals.

Asynchronous satellite: a satellite in a non-geostationary orbit; i.e., one that moves across the heavens as viewed from the earth.

AUTODIN: Automatic Digital Network (DCS).

AUTOSEVOCOM: Automatic Secure Voice System (DCS).

AUTOVON: Automatic Voice Network (DCS).

baud: a unit of signalling speed mathematically equal to the reciprocal of the duration of the shortest signal element to be transmitted.

bit: a unit of information content.

bps: bits per second.

CCITT: *Comité Consultatif International Télégraphique et Téléphonique*; the international organization responsible for development of communications standards.

circuit switching: the interconnection of communications users by means of tandem electrical connection of circuits between them in real-time.

compandor: an electronic device which improves signal-to-noise ratio by compressing the dynamic range of speech prior to transmission and expanding it back to its original range before connection to the receiving subscriber.

DATRAN: Data Transmission Company; a United States common carrier in the communications field.

db: decibel.

dbm: decibels referred to 1 milliwatt.

DCA: Defense Communications Agency.

DCS: Defense Communications System.

DCS backbone: that portion of the DCS which interconnects nodes or subscriber entry points; it does not include subscriber terminal equipment.

delta modulation: an adaptive form of PCM which reduces the data rate of digitized speech by encoding the difference between successive voice samples in accordance with a predetermined characteristic.

digital signal: an electrical signal which can assume only certain discrete, predefined levels, moving from one level to another discontinuously in time. (Theoretically, the transition between levels is instantaneous.) Mathematically, a multiform function of time.

digital system *or* Digital Communications System: a communications system which conveys information by altering the magnitude, position, width, or coded relationship between digital signals, and which provides for the routing of these information-bearing signals. As used

in this paper, a digital system does not involve conversion of the signal to analog form at any point.

gHz: gigahertz, 10^9 hertz (formerly gigcycles/second, also kilomegacycles/second; both terms now obsolete).

GPO: General Post Office (Great Britain); operator of the telephone system in that country.

GTE: General Telephone and Electronics; second-largest telephone common carrier in the United States.

hand-over: as applied to communications satellites, the procedure of changing from one satellite to another at a ground station. This consumes time, as the earth station antennas must physically be shifted from tracking the "old" satellite to tracking the "new" satellite.

IC: integrated circuit.

kb/sec: kilobits per second; 10^3 bits per second.

kHz: kilohertz, 10^3 hertz (formerly kilocycles/second; now obsolete).

LSI: Large Scale Integration; the process by which entire circuit functions and functional groups of circuits are fabricated on a single integrated circuit chip.

mb/sec: megabits per second; 10^6 bits per second.

MCI: Microwave Communications, Inc.; a United State communications common carrier.

message switching: the transfer of information between users by sequential routing of messages between nodes until they are transferred from originator to recipient; not a real-time system.

mHz: megahertz, 10^6 hertz (formerly megacycles/second; now obsolete).

msec: millisecond(s); 1/1000 of a second.

NRZ: non-return to zero; describes a digital code which can change states without passing first through the zero state.

packet switching: a means of data traffic routing; fully described in Chapter V.

PAR: pulse-to-average ratio; a method of determining the quality of a communications circuit.

Pierce Loop: a type of data network; fully described in chapter V.

PTT: *Postes, Téléphones, et Télégraphes*, the French phrase describing the national communications authorities in most of the world except the United States and Canada.

RF: radio frequency; those frequencies from approximately 100 kHz to the near infrared.

Synchronous satellite: a satellite in a geostationary orbit; i.e., one that does not appear to move as viewed from earth.

VF: voice frequency (approximately 200 to 3400 Hz).

μsec: microsecond(s); 1/1,000,000 of a second.

BIBLIOGRAPHY

BIBLIOGRAPHY

Government Publications

Defense Communications Agency. *System Description: DCS-AUTODIN*. DCA Circular 370-D95-1. November 1973.

________. *Operational Direction and Status Reporting for the Defense Communications System*. DCA Circular 310-55-1. July 1972.

________. *Communications Requirements: Defense Communications System Management Thresholds (MT)*. DCA Circular 310-130-2. 7 July 1970.

Department of the Army. *Vietnam Studies: Communications-Electronics, 1962-1970*. By T. M. Rienzi. 1972.

U.S. Congress. House. Committee on Armed Services. *Review of Department of Defense Worldwide Communications, Phase I* [*Mollohan Report*]. Armed Services Investigating Subcommittee report, 92d Cong., 1st sess. 10 May 1971.

U.S. Army Electronics Command. *Performance Rating for Analogue and Digital Modulation Methods*. Research and Development Technical Report ECOM-3269. By F. S. Gutleber. April 1970.

________. *Suitability of Existing Army Communications for Data Transmission*. Technical Report ECOM-2801. By J. R. Tucker and A. H. Gillette. Fort Monmouth, N. J., January 1967.

U.S. Army Signal School. *Data Communications*. Special Text. 1970.

________. *Satellite Communications*. Special correspondence course. 1968.

U.S. Army Test & Evaluation Command. *Data Transmission Equipment*. MTP 6-2-065. 25 September 1969.

Books

Bennett, W. R., and J. R. Davey. *Data Transmission.* New York: McGraw-Hill, 1965.

Brenner, E., and M. Javid. *Analysis of Electric Circuits.* New York: McGraw-Hill, 1959.

Hamsher, D. H. *Communication System Engineering Handbook.* New York: McGraw-Hill, 1967.

Karakash, J. J. *Transmission Lines and Filter Networks.* New York: Macmillan, 1950.

Martin, J. *Systems Analysis for Data Transmission.* Englewood Cliffs, N. J.: Prentice-Hall, 1972.

Ramo, S., and J. R. Whinnery. *Fields and Waves in Modern Radio.* 2d ed. New York: Wiley, 1953.

Terman, Frederick E. *Electronic and Radio Engineering.* 4th ed. New York: McGraw-Hill, 1955.

Reports

Abramson, N. *The Aloha System--Another Alternative for Computer Communications.* University of Hawaii Technical Report B70-1. April 1970.

All-Digital 1980 DCS Study, Phase I, Task IV A, Final Report (U). Prepared by IBM and Philco-Ford under Contract DCA 100-70-C-0001. November 1970.

COMSAT Annual Report to the President and the Congress, 1973. Washington, 31 July 1973.

Digital Data System Channel Interface Specifications (Preliminary). Bell System Publication 41021. March 1973.

General Dynamics Electronics Division. *Study of Electronic Handling of Mail: Transmission Methods.* Report R-69-046-5. Prepared for Bureau of Research and Engineering, Post Office Department Washington. July 1970.

AD-A089 534 ARMY COMMAND AND GENERAL STAFF COLL FORT LEAVENWORTH KS F/G 17/2
CONVERSION OF THE DEFENSE COMMUNICATIONS SYSTEM FROM ANALOG TO --ETC(U)
JUN 74 R A STANLEY

UNCLASSIFIED SBIE-AD-E750 001 NL

3 of 3

AD A
089534

END
DATE
FILMED
10-80
DTIC

Investigation of TDM/PCM Multiple-Access Techniques for Satellite Communications. General Post Office Research Department Report No. 64. London, January 1969.

Leifer, P. L., and D. P. Campbell. "Converter, Analog-Digital, CV-3034/G. Fourth Quarterly Report, Report #2304-2508-32. Contract DAAB07-72-C-0093. Philadelphia: General Atronics Corporation, January 1973.

"9101C PCM Repeatered Line Equipment." *GTE Lenkurt Practices*, Issue 2, November 1972. (Section 342-910-103.)

Wentz, L. K. *Overseas AUTOVON Anomalies*. MTR-2761. Boston: MITRE Corporation, 8 January 1974.

Witt, R. P., S. Barosin, and R. A. Culpon. *An Economic Comparison of Analog vs. Digital-Based Common User Communications*. MTR-1930. Contract F19(628)-68-C-0365. Bedford, Mass.: MITRE Corporation, November 1970.

Witt, R. P., S. Barosin, and K. D. Kasparian. *Air Force Planning for Digital Communications*. MTR-1803. Contract F19(628)-68-C-0365. Bedford, Mass.: MITRE Corporation, August 1970.

Articles and Periodicals

Altshuler, E. E., V. J. Falcone, and K. N. Wulfsberg. "Atmospheric Effects on Propagation at Millimeter Wavelengths." *IEEE Spectrum*, V, No. 7 (July 1968), 83-90.

Anderson, R. R., W. R. Bennett, J. R. Davey, and J. Salz. "Differential Detection of Binary FM." *Bell System Technical Journal*, XLIV, No. 1 (January 1965), 111-59.

Andrews, M. C. "New Data Networks in Europe." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1369.

"AT&T's Rate Plans Jolt the Competition." *Business Week*, 3 March 1973, p. 26.

Auerbach, I. L. "Technology and the Future." *Data Systems*, January 1971, pp. 12-13, summarized in *Data Processing Digest*, XVII, No. 6 (June 1971), 1-2.

Avi-Itzhak, B. "Heavy Traffic Characteristics of a Circular Data Network." *Bell System Technical Journal*, L, No. 8 (October 1971), 2521-50.

Baird, Jack R. "An Optical Data Link for Remote Terminals." *Datamation*, January 1970, pp. 125-26.

Balkovic, M. D., *et al.* "High-Speed Voiceband Data Transmission Performance on the Switched Telecommunications Network." *Bell System Technical Journal*, L, No. 4 (April 1971), 1349-84.

Bayless, J. W., S. J. Campanella, and A. J. Goldberg. "Voice Signals: Bit-by-Bit." *IEEE Spectrum*, X, No. 10 (October 1973), 28-36.

Bender, E. C., and R. D. Howson. "Wideband Data Service." *Bell System Technical Journal*, L, No. 2 (February 1971), 667-82.

Berger, U. S. "The Old TD2 Becomes the New TD2C." *Bell Laboratories Record*, LI, No. 9 (October 1973), 279-84.

Blair, R. W., and R. S. Burnell. "Monitors Take the Pulse of T-1 Transmission Lines." *Bell Laboratories Record*, LI, No. 2 (February 1973), 55-60.

Bobsin, J. H., and L. E. Forman. "The T2 Digital Line--Extending the Digital Network." *Bell Laboratories Record*, LI, No. 8 (September 1973), 239-43.

Bowers, R., and J. Frey. "Technology Assessment and Microwave Diodes." *Scientific American*, CCXXVI, No. 2 (February 1972), 13-21.

Broderick, C. W. "A Digital Transmission System for TD-2 Radio." *Bell System Technical Journal*, L, No. 2 (February 1971), 481-500

Burwell, G. H., *et al.* "A Portable Test Set for Detecting Errors on T-1 Lines." *Bell Laboratories Record*, L, No. 1 (January 1972), 14-17.

Childers, A. A. "The Introduction of Digital Transmission into the DCS." *Signal*, XXV, No. 7 (March 1971), 42-44.

Coker, C. H. "An Experimental Interconnection of Computers Through a Loop Transmission System." *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1167-76.

Combs, B. "Tymnet: A Distributed Network." *Datamation*, July 1973, pp. 40-43.

"Concentrators and Multiplexers." *GTE Lenkurt Demodulator*, August 1973, entire issue.

Cox, J. E. "Western Union Digital Services." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1350-57.

"Crosstalk." *Lenkurt Demodulator*, November 1960, entire issue.

Cunningham, P. B., *et al.* "A New T-1 Regenerative Repeater." *Bell Laboratories Record*, XLIX, No. 4 (April 1971), 109-114.

"Data Timing for HF Radio Transmission." *Lenkurt Demodulator*, February 1964, entire issue.

"Development of Multiplex Telephony, The." *Lenkurt Demodulator*, XIV, No. 12 (December 1965), entire issue.

"Difficult Problem of Exchange Trunk Carrier, The." *Lenkurt Demodulator*, October 1960, entire issue.

Duttweiler, D. L. "Waiting-Time Jitter." *Bell System Technical Journal*, LI, No. 1 (January 1972), 165-207.

Elliott, E. O. "A Model of the Switched Telephone Network for Data Communications." *Bell System Technical Journal*, XLIV, No. 1 (January 1965), 89-109.

Falk, Howard. "Technology '74: Communications and Microwaves." *IEEE Spectrum*, XI, No. 1 (January 1974), 34-36.

________. "Technology '74: Data Communications." *IEEE Spectrum*, XI, No. 1 (January 1974), 36-39.

Fano, R. M. "On the Social Role of Computer Communications." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1249-53.

Fennick, J. H. "Transmission Objectives for Voiceband Data." *Bell Laboratories Record*, L, No. 1 (January 1972), 18-21.

"Five-Year Plan Goal: Improved Mechanisms for IC Reliability." *Electronics*, XLVII, No. 1 (10 January 1974), 38-40.

Fleming, H. C., and R. M. Hutchinson, Jr. "Low-Speed Data Transmission Performance on the Switched Telecommunications Network." *Bell System Technical Journal*, L, No. 4 (April 1971), 1385-1405.

Frank, H., and W. Chou. "Topological Optimization of Computer Networks." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1385-97.

Franklin, R. H. "Compatibility in World Communications." *IEEE Spectrum*, III, No. 10 (October 1966), 73-77.

________, and H. B. Law. "Trends in Digital Communications by Wire." *IEEE Spectrum*, III, No. 11 (November 1966), 52-58.

Fultz, K. E., and D. B. Penick. "The T1 Carrier System." *Bell System Technical Journal*, XLIV, No. 7 (September 1965), 1405-1451.

Geigel, Al. "Monitoring the Performance of Digital Multiplex Circuits." *Bell Laboratories Record*, XLIX, No. 7 (August 1971), 198-202.

Glance, B. "Power Spectra of Multilevel Digital Phase-Modulated Signals." *Bell System Technical Journal*, L, No. 9 (November 1971), 2857-78.

"Glossary of Signaling Terms, A." *GTE Lenkurt Demodulator*, April 1974, entire issue.

Gould, G. T., Jr. "Military Communications in the Information Age." *Signal*, XXV, No. 1 (September 1970), 11-12.

Gould, R. G., and G. K. Helder. "Transmission Delay and Echo Suppression." *IEEE Spectrum*, VII, No. 4 (April 1970), 47-54.

Gunn, J. F., *et al.* "Mastergroup Digital Transmission on Modern Coaxial Systems." *Bell System Technical Journal*, L, No. 2 (February 1971), 501-520.

Higgins, W. H. C. "Major Trends in Switching." *Bell Laboratories Record*, L, No. 9 (October 1972), 270-80.

Hill, F. S., Jr. "The Computation of Error Probability for Digital Transmission." *Bell System Technical Journal*, L, No. 6 (July-August 1971), 2055-77.

Ho, E. Y., and D. A. Spaulding. "Data Transmission Performance in the Presence of Carrier Phase Jitter and Gaussian Noise." *Bell System Technical Journal*, LI, No. 8 (October 1972), 1927-31.

James, R. T., and P. E. Muench. "AT&T Facilities and Services." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1342-49.

Johnson, G. D. "No. 4 ESS--Long Distance Switching for the Future." *Bell Laboratories Record*, LI, No. 8 (September 1973), 226-32.

Kahl, H. "Evolution of Wideband Data Services." *Bell Laboratories Record*, XLIX, No. 8 (September 1971), 244-50.

"Keep an Eye on Bell's ISAL [Information System Access Lines] Tariff." *Business Automation* (News Analysis Edition), 15 February 1971, pp. 6-7 & 12.

Kessler, J. E. "The Transmission Performance of Bell System Toll Connecting Trunks." *Bell System Technical Journal*, L, No. 8 (October 1971), 2741-78.

Komerth, K. L., and B. R. Shah. "Optical Transmission Utilizing Injection Light Sources." *IEEE Spectrum*, VII, No. 9 (September 1970), 37-46.

Kretzmer, E. R. "The New Look in Data Communications." *Bell Laboratories Record*, LI, No. 9 (October 1973), 258-65.

Kropfl, W. J. "An Experimental Data Block Switching System." *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1147-66.

Lapidus, Gerald. "New Solid-State Circuits." *IEEE Spectrum*, XI, No. 1 (January 1974), 99.

"Largest Civilian Government Communications Procurement in History, The." *Data Channels*, I, No. 2 (March 1974), 4-5.

Law, P. E. "Communications System Planning." *Signal*, XXVII, No. 4 (December 1972), 22-26.

"Load Capacity of High-Density Multiplex Systems." *Lenkurt Demodulator*, XIV, No. 3 (March 1965), entire issue.

"Low-Loss Coupler Feeds 20 Terminals." *Electronics*, XLVII, No. 6 (21 March 1974), 34.

Lucky, R. W. "Automatic Equalization for Data Communication." *Bell System Technical Journal*, XLIV, No. 4 (April 1965), 547-88.

________, and H. R. Rudin. "An Automatic Equalizer for General-Purpose Communications Channels." *Bell System Technical Journal*, XLVI, No. 9 (November 1967), 2179-2208.

McKay, R. G. "Digital Communications--A Tutorial." *Bell Laboratories Record*, XLIX, No. 9 (October 1971), 278-84.

McPherson, J. C. "Data Communication Requirements of Computer Systems." *IEEE Spectrum*, IV, No. 12 (December 1967), 42-45.

Mathison, S. L., and P. M. Walker. "Regulatory and Economic Issues in Computer Communications." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1254-72.

Mayo, J. S. "Experimental 224 Mb/s PCM Terminals." *Bell System Technical Journal*, XLIV, No. 9 (November 1965), 1813-41.

"Methods for Transmitting Data Faster." *Lenkurt Demodulator*, IX, No. 3 (March 1960), entire issue.

"Millimeter Waveguide To Be Tested in West Germany." *Electronics*, XLVII, No. 4 (21 February 1974), 55.

Molnar, J. P. "The Telephone Plant of the 1970's." *Bell Laboratories Record*, XLIX, No. 1 (January 1971), 2-12.

________. "A Telecommunications Superhighway for the 1970's." *Signal*, XXV, No. 1 (September 1970), 44-47.

"Multiplexing and Modulation in Carrier Telephone Systems." *Lenkurt Demodulator*, VIII, No. 1 (January 1959), entire issue.

Nelson, F. B. "Data Transmission in Transition." *Datamation*, 1 October 1971, pp. 20-24.

"New Chief of the FCC's Common Carrier Bureau Backs Competition in Telecommunications." *Electronics*, XLVII, No. 2 (24 January 1974), 31.

Nordling, K. I. "Analysis of Common Carrier Tariff Rates." *Datamation*, 15 April 1971, pp. 28-35.

Norman, Janet L. "Reducing Telephone Network Errors." *Datamation*, 1 October 1971, pp. 24-28 & 30-31.

Noyce, R. N. "Making Integrated Electronics Technology Work." *IEEE Spectrum*, V, No. 5 (May 1968), 63-66.

Nyquist, H. "Certain Topics in Telegraph Transmission Theory." *Transactions of the AIEE*, XLVII, No. 4 (April 1928), 617-44.

"PCM Cable Considerations." *GTE Lenkurt Demodulator*, June 1973, entire issue.

"PCM-FDM Compatibility." *GTE Lenkurt Demodulator*, July, August, and September 1971, entire issues.

"PCM Field Study." *GTE Lenkurt Demodulator*, March 1974, entire issue.

Peters, R. A., and K. M. Simpson. "Eurodata: Data Communications in Europe, 1972-1985." *Datamation*, December 1973, pp. 76-80.

"Phase Jitter." *GTE Lenkurt Demodulator*, December 1972, entire issue.

Pierce, J. R. "Network for Block Switching of Data." *Bell System Technical Journal*, LI, No. 6 (July-August 1972), 1133-46.

________. "Some Practical Aspects of Digital Transmission." *IEEE Spectrum*, V, No. 11 (November 1968), 63-70.

"Requirements of a Human Communications Channel." *Lenkurt Demodulator*, September 1965, entire issue.

Ristenblatt, M. P. "Alternatives in Digital Communications." *Proceedings of the IEEE*, LXI, No. 6 (June 1973), 703-721.

Roberts, L. G. "Data by the Packet." *IEEE Spectrum*, XI, No. 2 (February 1974), 46-51.

Rudin, H., Jr. "Data Transmission: A Direction for Future Development." *IEEE Spectrum*, VII, No. 2 (February 1970), 79-85.

________. "Automatic Equalization Using Transversal Filters." *IEEE Spectrum*, IV, No. 1 (January 1967), 53-59.

Schindler, H. R. "Delta Modulation." *IEEE Spectrum*, VII, No. 10 (October 1970), 69-78.

Schulke, H. A., Jr. "The DCS Today and How It Operates." *Signal*, XXV, No. 11 (July 1971), 22-28.

Shannon, C. E. "A Mathematical Theory of Communication." *Bell System Technical Journal*, XXVII: No. 6 (July 1948), 379-423; and No. 8 (October 1948), 623-56.

Smith, L. B. "Communicating With Computers Over Data Concentration Systems." *Bell Laboratories Record*, L, No. 5 (May 1972), 144-49.

"Subscriber Carrier." *Lenkurt Demodulator*, March 1970, entire issue.

"Theory and Use of Compandors in Voice Transmission Systems, The." *Lenkurt Demodulator*, October 1964, entire issue.

Thiel, F. L., and W. B. Bielawski. "Optical Wave-Guides Look Brighter Than Ever." *Electronics*, XLVII, No. 6 (21 March 1974), 89-96.

Thompson, M. J. "Transmitting Digital Signals on Analog Radio Links." *Bell Laboratories Record*, XLIX, No. 4 (April 1971), 102-108.

"Tomorrow's Phone Service--Why It Will Cost More." *U.S. News & World Report*, 9 April 1973, pp. 57-62.

Travis, L. F., and R. E. Yeager. "Wideband Data on T1 Carrier." *Bell System Technical Journal*, XLIV, No. 8 (October 1965), 1567-1604.

"U.S. Markets, 1974: Communications." *Electronics*, XLVII, No. 1 (10 January 1974), 107-108.

Watson, G. F. "Evolution From Analog To Digital." *Bell Laboratories Record*, LI, No. 8 (September 1973), inside front cover.

Weiner, J. H. "U.S. Military Worldwide Communications." *IEEE Spectrum*, III, No. 10 (October 1966), 80-81.

Welber, I. "A Commitment." *Bell Laboratories Record*, L, No. 1 (January 1972), inside front cover.

White, G. "Optical Modulation at High Information Rates." *Bell System Technical Journal*, L, No. 8 (October 1971), 2607-2645.

________. "A One-Gigabit-Per-Second Optical PCM Communications System." *Proceedings of the IEEE*, LVIII, No. 10 (October 1970), 1779-80.

Worley, Allen R. "The DATRAN System." *Proceedings of the IEEE*, LX, No. 11 (November 1972), 1357-68.

Unpublished Material

Baran, Paul. "LSI: The Basic Module for New Communication Networks." Paper presented before IEEE Computer Group Conference on LSI, Los Angeles, 27-28 June 1968.

Duff, T. O. "The Defense Communications Agency." U.S. Air Force Air War College Report #4564, April 1972.

Mills, F. L. "Centralization of the U.S. Army Telecommunications Computer Programming Effort." U.S. Army Command and General Staff College MMAS thesis, 1971.